

THE SOUND ENGINEERING MAGAZINE

MARCH 1981

\$1.95

db

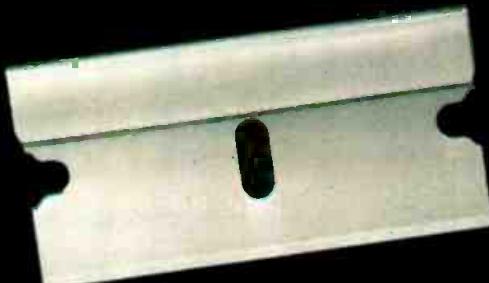


Our Digital Editor is a Five Cent Option.

All other digital 2-channel audio recorders require special editing equipment which could raise your initial investment by as much as \$25,000 or more. The Mitsubishi X80 Digital 2-channel Audio Recorders require only a standard razor blade to edit, even within the music portion.

In designing the Mitsubishi X80 series digital audio recorders, we've made the human engineering process as simple as a standard analog system but, recording with the X80 captures and plays back true sound. The depth. The presence. The richness. We call it Three-Dimensional Sound.

When you hear music captured on the Mitsubishi X80, it's like being



right in the booth with the players. There's no hiss. No print through. No drop outs. No deterioration through generations. The Mitsubishi X80 (portable) and the X80A (console) are the culmination of years of refining and perfection and are available for delivery right now.

Our price is as remarkable as our sound; approximately one-half the price of other 2-channel digital recording systems.

The Mitsubishi X80 series was designed with you in mind.

Who needs digital?
The Answer: You.

Digital audio recording is not a fad, but rather the inevitable future of recorded sound.

The Digital Audio Disk Standardization Committee, in Tokyo, has put its seal of

approval on a compact disk format for home digital audio recorders that uses optical laser technology to play-back digital audio disks. Through this laser signal, sound can be reproduced much cleaner and more realistically than ever before.

Digital audio recording is as big a sound breakthrough to analog recording as stereo recording was to the Hi-fi industry. Digital audio disks and software will soon be a reality in Japan and will be available in the United States in 1982.

Get in on the ground floor of digital audio recording with the pioneers in the field, Mitsubishi. The X80/X80A are only one part of a whole digital system we've designed to meet the changing and demanding audio needs of the music industry. Our systems work now and will keep working for you.

For more information on the Mitsubishi X80 2-channel Digital Audio Recorders and Mitsubishi Digital Audio Systems call us at 800-323-4216 (outside Ill.) or 312-982-9282 (within Ill.)

EXPERIENCE THE 3rd. Dimension of SOUND.

THE CONSOLE TYPE X80A



THE TRANSPORTABLE X80

X80 X80A

**2-Channel
Digital Audio Recorder**

 **MITSUBISHI**
DIGITAL AUDIO SYSTEMS

Circle 10 on Reader Service Card

Larry Zide
Publisher

John M. Woram
Editor

Mark B. Waldstein
Associate Editor

Kathy Lee
Advertising Production & Layout

Carol Vitelli
Classified Advertising

Lydia Anderson
Book Sales

Eloise Beach
Circulation Manager

Bob Laurie
Art Director

Crescent Art Service
Graphics

Spartan Phototype Co.
Typography

Sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75207
1949 Stemmons Freeway, Suite 670
(214) 742-2066

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East, Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive, Suite 210
(713) 988-5005

Los Angeles Area
Glendale, California 91204
424 West Colorado Street, Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
(503) 640-2011

San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
(415) 653-2122

Sagamore Publishing Co., Inc.
New York
Plainview, NY 11803
1120 Old Country Rd.
(516) 433-6530

ABOUT THE COVER—

• In the Glendale, California, R & D studios of Yamaha International, a mostly-electronic selection of the company's M1 products introduces us to this month's topic of "Music in the Studio: Electronic." Reading clockwise from lower-left foreground: CP80 Electric Piano, CS40M Programmable Memory Synthesizer, B100-115SE Bass Amp, BB-800 Bass, C-7 Piano, SBG-2000 Guitar, G100-112 Guitar Amp, 9000 series Drums (in reflection), GS-1 Digital Synthesizer, and GS-1 Programmer. The programmer has four CRT displays to simultaneously represent its four oscillator banks. Photo courtesy of Thom Elder.



THE SOUND ENGINEERING MAGAZINE

MARCH 1981 NUMBER 15, VOLUME 3

EDITORIAL	34
VOCODERS	35
Felix Visser	
REPRODUCING ELECTRONIC MUSIC IN THE CONTROL ROOM	41
Michael Rettinger	
METRIC MUSIC	44
Hal Lion and Jim Fox	
ZOOP, BEEP, BRAP, BROOP	46
Ralph Hodges	
DIRECTORY OF ELECTRONIC MUSIC SYNTHESIZER MANUFACTURERS	50
A db CONVENTION REPORT: THE 67th AES CONVENTION	52
LETTERS	6
CALENDAR	14
DIGITAL AUDIO	16
Barry Blesser	
THEORY AND PRACTICE	20
Norman H. Crowhurst	
SOUND WITH IMAGES	26
Martin Dickstein	
NEW PRODUCTS AND SERVICES	32
CLASSIFIED	54
PEOPLE, PLACES, HAPPENINGS	56



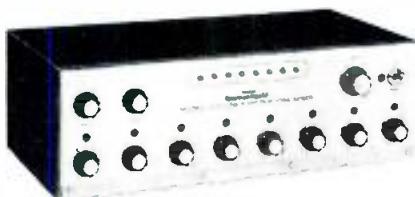
is listed in **Current Contents: Engineering and Technology**

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright 1981 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 233-6530. db is published for those individuals and firms in professional audio recording, broadcast, audio visual, sound reinforcement consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are \$12.00 per year (\$24.00 per year outside U.S. Possessions and Mexico, \$13.00 per year Canada; in U.S. funds. Single copies are \$1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Old Saybrook, Connecticut 06475.



PURE PRO audio design

introduces the remarkable
3535 LOG-8 Microphone Mixer



with *Voice-Entrée* option

Permits the activation and simultaneous use of multiple microphones without feedback and does the job automatically—no operator, no manual adjustments required...

ASK FOR DESCRIPTIVE FOLDER

**LOOK TO RAULAND FOR EVERYTHING
IN PROFESSIONAL SOUND**



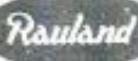
Spectrum-Master Equalization (Patented): Most complete line available. Includes 1/3- and 1-octave models, test set, and tunable notch filter.

Spectrum-Master In-Wall Amplifiers: Available in 35-watt, 60-watt, 100-watt outputs. Each with built-in 1-octave equalizer and D.R.E.

Spectrum-Master Amplifiers: Incomparable Model DX (5-year warranty) and TAX amplifiers, from 70 to 250 watts RMS to meet any professional audio requirement.

Spectrum Series Speakers: Professional quality two-way speaker systems embodying superior components in advanced system designs

WRITE FOR TECHNICAL BULLETINS



*The Quality Name in
Professional Audio*

RAULAND-BORG CORPORATION
3535 W. Addison St., Dept. N., Chicago, Ill. 60618

Circle 25 on Reader Service Card

db Letters

Index of Advertisers

Ampex	29
Audio-Technica	11
Bose	9
Cetec Vega	37
Crown	13
Eastman School of Music	28
ELAR Publishing	33
Electro-Voice	15
ESE	49
Garner Industries	20
JBL	19
MAC 81	50
Mike Shop	12
Milab	22
Mitsubishi	Cover II
R. K. Morrison Illust. Mats.	16
Museum of Archives & Magnetic Recording	32
Neve	Cover IV
Nortronics	23
Orban Associates	24
Pioneer	25
Polyline	28
Pro Audio Seattle	14
Rauland-Borg	6
Sescom	16
Sennheiser Electronics	18
Shure	7
Sound Technology	30, 31
Soundcraftsmen	43
Standard Tape Lab	45
Swintek	18
Teac	21
Telex	8
Telex Turner	10
Tektronix	Cover III
Tentel	44
Tweed Audio USA	17
Valley People	27
Waters Manufacturing	26
White Instruments	12

Coming Next Month

- In April, it's "Music in the Studio: Acoustic," in which we continue our examination of the musical side of the recording industry (it's not *all* electronic you know). Plus more on the analog/digital controversy, and a little broadcast audio thrown in to mark the upcoming NAB convention.

PRO MASTER™: the uncomplicated way to professional sound



fact:

Shure introduces a new sound system in the tradition of the famous Shure VOCAL MASTER®—with state-of-the-art features and truly outstanding performance: the PRO MASTER.

Designed especially for churches, synagogues, schools, hotels, motels, restaurants, and auditoriums, the new system is incredibly easy to set up and operate—even by first-time users! Yet, veteran soundmen agree it delivers a new order of audio control, clarity, and purity, PLUS improved PRO MASTER reliability.

Put it together: the new 706 Power Console featuring a 200-watt amplifier, and a pair of the new 709 Loudspeakers. An outstanding combination at an unbeatable price. Great sound was never this easy!

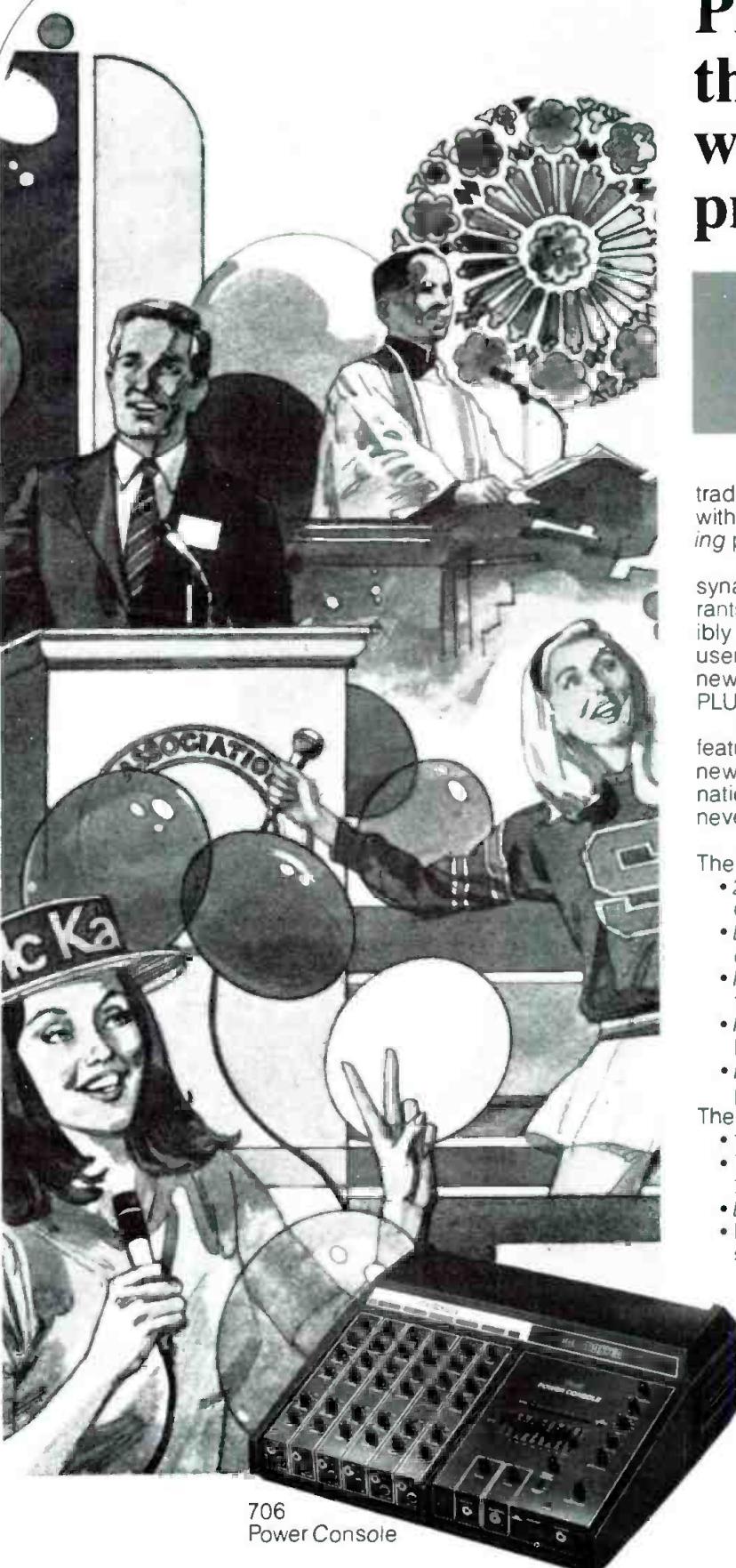
The PRO MASTER 706 Power Console features:

- 200-watt amplifier, with 10-band graphic equalizer
- Eight-channel mixer, with logically located controls
- FEEDBACK FINDER™ to help eliminate feedback fast
- PATCH BLOCK™ rear panel simplifies hookup
- LED status indicators alert you to potential problems

The PRO MASTER 709 Loudspeaker features:

- 150-watt power rating
- Three piezoelectric tweeters and 15-inch woofer
- Easy-to-carry enclosure with plenty of output
- Plus it's super portable. the console and two speakers fit in most subcompact cars!

See it today at your Shure dealer, or write Shure for Brochure AL647.



706
Power Console

 **SHURE**®

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204
In Canada: A. C. Simmonds & Sons Limited
Manufacturers of high fidelity components, microphones,
sound systems and related circuitry.



709
Loudspeaker System



All Wireless Microphones Are Not Created Equal

This One is a
Telex

Wireless mics aren't new, and sometimes it seems as if all systems are basically the same. However, Telex and its Turner and Hy-Gain divisions have combined their 100 years of cumulative experience in microphone, antenna and rf development to produce a DUAL DIVERSITY WIRELESS SYSTEM THAT COSTS AS LITTLE AS SINGLE ANTENNA INSTALLATIONS. The FM receiver can be operated with one or two antennas. When two antennas are used, a unique automatic phase summation network (patent applied for) provides superb dual diversity reception.

The Telex wireless sounds as good as a hard wired mic, offers plenty of options and is economically priced. If you're interested in a wireless system that is *more* than equal—write us today for full specifications.

Quality products for the audio professional



TELEX COMMUNICATIONS, INC.

9600 Aldrich Ave. So., Minneapolis, MN 55420 U.S.A.
Europe 22, rue de la Légion-d'Honneur, 93200 St. Denis, France

Circle 33 on Reader Service Card

I do hope that you can help us, and perhaps if you know any company who has taken over Garron's business, you will let us know so that we can try yet another tack. Perhaps you are aware of another broadcasting service in the USA who has used these machines who may be able to lay their hands on an old manual.

We look forward to hearing from you. With many thanks, and in anticipation of your valued help, we remain,

DAVID GARFIELD
Station General Manager
London Hospitals' Broadcasting

db replies:

How about it readers? Can anyone help Mr. Garfield?

WHAT'S A DYNAMIC CONDENSER MICROPHONE?

TO THE EDITOR:

I would like to correct some misprints in my article, "Hum, Pop, Thump, and Other Microphone Noises" in the December, 1980 issue of db.

In FIGURES 7 and 9, the frequency scale shown is ten times too large; that is, 1 kHz should be 100 Hz, 100 Hz should be 10 Hz, and so on. In FIGURE 7, the shock-mount resonant frequency should be 9 Hz, not 90 Hz.

The last sentence in the article should read, "For minimum self-noise: Choose high-sensitivity dynamic *or* condenser microphones designed for low-noise performance."

BRUCE A. BARTLETT
Senior Development Engineer
Electroacoustical Development

db replies:

Sorry 'bout that! Your graphs ran afoul of our art department. And, by leaving out the word "or," we "invented" a new kind of microphone.

MORE ON STEREO MIKING

TO THE EDITOR:

In Thomas Ammons' letter (db December 1980), he asked what would be the stereo (X-Y) microphone patterns resulting from using M-S technique with the M set in omni or figure-8. Since neither your reply nor the article to which you referred him actually answered the question, I'm enclosing a diagram which shows the gamut of M-S patterns, with the equivalent X-Y patterns to the left.

It might be noted that using a figure-8 for the M, as well as the S (which results in crossed figure-8s in X-Y), was preferred by Blumlein, the inventor of M-S technique. I've found this useful in



"The closest damn thing."

We were recently lucky enough to catch Doug Kershaw on tour. After his show at Harrah's at Lake Tahoe he talked to us about his new Bose® sound system, which consists of four Bose 802 speakers, a Bose PM-2 Powermixer, and a Bose 1800 power amplifier. He's been using the system to amplify his electric fiddle, squeeze box, electric guitar, and vocals.

Q: Doug, you've been playing for a long time. I'll bet you've tried a lot of different kinds of sound gear, haven't you?

Kershaw: Yes, I've used lots of different things and I've spent a lot of time developing my sound. Even then, I could never quite get what I was looking for. But my new Bose system is the closest thing to what I want. The closest damn thing.

Q: What differences have you noticed since you started using the Bose system?

Kershaw: For one thing, it doesn't hurt my ears. You know, I've used some big speakers that have almost busted my ears. I've even put my foot through a few of them. But this is a true sound. It sounds just like my fiddle, no matter how loud I turn it up.

Q: Have you found that you have changed your playing in any way because of how the 802s perform for you?

Kershaw: The attack is easier. It's just easier.

If a Bose system makes it easier for Doug Kershaw, you might find that it can do the same for you. Visit a Bose Pro Products dealer soon and try one out.

Bose Corporation, Dept. SE
The Mountain
Framingham, MA 01701

Please send me a copy of the Bose Professional Products catalog and a complete dealer list.

Name _____

Street _____

City _____

State _____ Zip _____

Phone () _____

Covered by patent rights issued and/or pending.
© Copyright 1981 Bose Corporation

BOSE®

Better sound through research.

Desk Paging?



Turner has More!

Turner desk paging microphones allow the audio professional more flexibility in choosing the right microphone for each installation. A minimum amount of electronic modification is required because Turner has engineered its products to meet virtually all applications. Twelve distinctly different microphones in two desk-top case styles. There is a quality Turner desk paging microphone with features to meet the following applications requirements:

- Omnidirectional • Low Impedance • Normally Open Switching
- Normally Closed Switching • High Impedance • Zone Paging
- Cardioid • Noise Cancelling • Press-To-Talk • Lift-To-Talk
- Amplified.

And, that's only the beginning. Turner has 13 other paging microphones in gooseneck, handheld and wall mount versions, as well as a full line of sound reinforcement microphones. Turner does have more, and now, with the additional product development strength of Telex Communications, Inc. there will be even more to come.

Quality Products for The Audio Professional



TELEX® TURNER®

TELEX COMMUNICATIONS, INC.

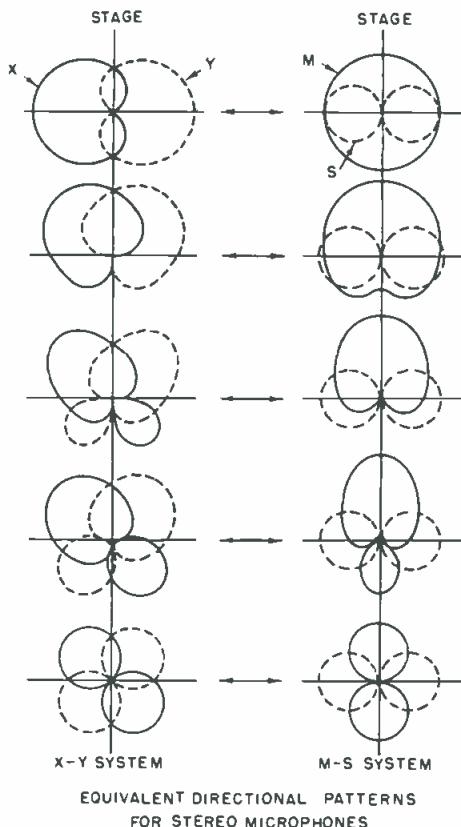
9600 ALDRICH AVE. SO. MINNEAPOLIS, MN 55420 U.S.A.
EUROPE: 22, rue de la Legion-d'Honneur, 93200 St. Denis, France

Circle 38 on Reader Service Card

orchestral recordings when one wants to reject some of the early reflections from the sides of the stage. At the other end of the scale, an omni M (or cardioids back-to-back in X-Y) offers an unusual stereo pick-up without the typical loss of off-axis sounds in mono.

Thanks for this opportunity to reply to Mr. Ammons.

LES STUCK
Hyde Street Studios
San Francisco



db replies:

And thanks for the additional information. Actually, our application note cited one of the classic M-S combinations. As reader Stuck correctly points out, there are many variations on the M-S theme, and each will produce a different pair of resultant X-Y patterns (and vice versa). In the accompanying diagram, note that the third example produces X-Y patterns that closely resemble supercardioids. On the other hand, two X-Y cardioids (example 2) create an M-S combination in which the M pattern is cardioid-like, but only down about 10 dB at 180 degrees. The combinations are endless, and given the high reader interest in stereo miking, we'll return to this subject in our next microphone issue (June, 1981).

Never Play Naked Records on the Air!

Your records need protection. From minor scratches, dust, dirt, and just too much handling, no matter what your program format.

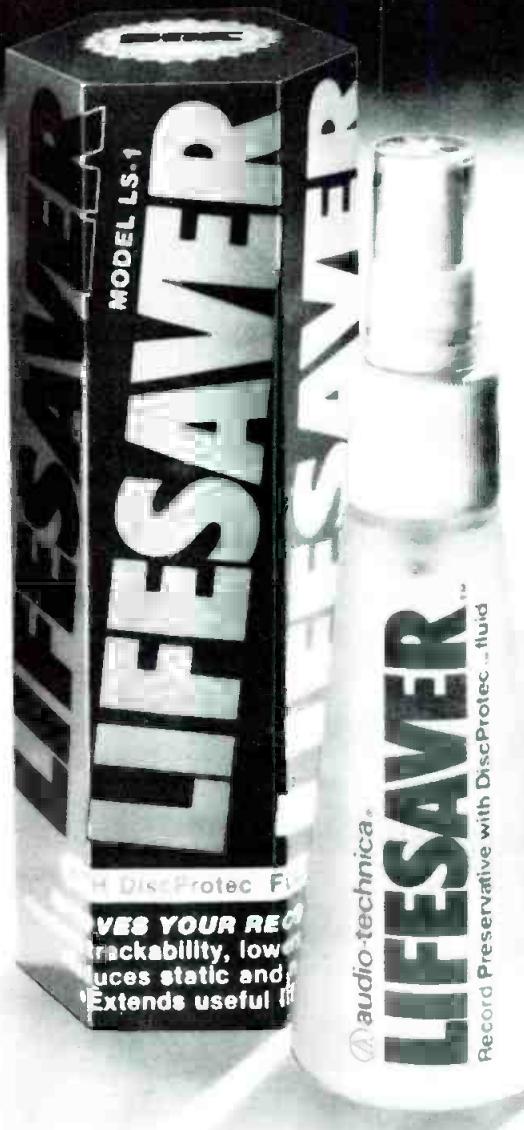
Introducing Lifesaver with Disc-Protect™ Formula. A one-step record preservative, anti-stat, and dry lubricant. One treatment lasts 50 plays or more (Len Feldman* proved Lifesaver lasts 100 plays, but we'll be conservative).

Records Sound Better Than New

From the very first audition, a Lifesaver-treated record will have less distortion than even a new untreated record. And 50 or even 100 plays later, it will still outperform that new record, with no build-up of record noise!

Fewer Repeats, Repeats, Repeats...

With most Lifesaver treatments, minor scratches and nicks play right through without repeats (no guarantee, but the odds are with us). So if you're into golden oldies, or vintage jazz, or classical music, you don't have to cross your fingers, or do those abrupt fades in the middle of the best part. Nor will you need extra copies of the records on your "heavy playlist."



A Sub-microscopic Coating

A quick "spritz," a gentle buffing, and you've dry-coated the record with an anti-stat/lubricant just microns thick. You can't see it even with a good microscope, you can't hear it, but you can feel the surface get slick and stay slick, and you'll notice no static cling of dust or record sleeves. Record cleaning is actually improved (Lifesaver is unaffected by any common wet or dry methods). And even years later, when you dig through your music library the Lifesaver-treated records will still be protected, still sound great.

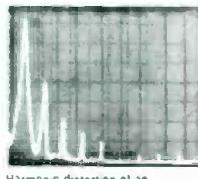
Save Records, Save Money

Get the dust, the static, and the minor scratches out of your music library today. Preserve the investment in your music and effects production library. Start using Lifesaver from Audio-Technica today. You'll sound better tomorrow! **AUDIO-TECHNICA U.S., INC.**, 1221 Commerce Drive, Stow, OH 44224, Dept. 31BD.

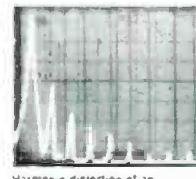


audio-technica

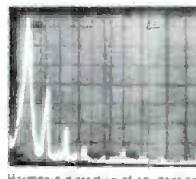
*Here's an excerpt from the Len Feldman report in *Audio Magazine*. We'll send you the full story with your order.



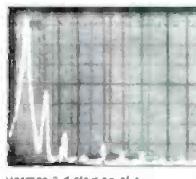
Harmonic distortion of an untreated disc during first playing.



Harmonic distortion of an untreated disc after 100 playings.



Harmonic distortion of an identical disc, first playing after LIFESAVER treatment. Distortion is immediately reduced.



Harmonic distortion of a LIFESAVER-treated disc after 100 playings. Distortion remains lower than a new, untreated disc.

SPECIAL INTRODUCTORY OFFER

To encourage you to prove the effectiveness of Lifesaver treatment, we'll send you a full-sized \$12.95 bottle to treat 60 sides for just \$5.00. Just send this ad (or a copy) and your letterhead. Just one sample to a station, please. And we'll tell you where you can get low-cost refills, plus see and hear other fine Audio-Technica products, including the latest microphones, headphones, other record care products, and the best cartridges and tone arms in the business! Send today.

mikes by mail? for less? why not!™

and much more!

The Mike Shop™ now sells audio equipment
as well as mikes by mail! for less!

Write or call us with your requirements or for our price sheet.



The Mike Shop™

PO Box 366A, Elmont, NY 11003 (516) 437-7925

A Division of Omnisound Ltd.

Circle 24 on Reader Service Card

TO THE EDITOR:

It was a real pleasure to read the article "Getting Down to 2 Tracks Fast..." by Ralph Hodges in your December 1980 issue. I didn't think there were any other old "dinosaurs" who would run around a studio with a finger stuck in one ear. Certainly, some of the modern mix-down methods are a marvel of modern technology, but the engineers who grew up on "fix-it in the mix" just don't have the skills to do it any other way.

Those of us who were around in the days of mono and helped to develop the original stereo techniques learned that the most important things to know were the subtle nuances of each microphone in the storage cupboard.

Thanks again for a most enjoyable article.

W. R. GRAHAM
Audio-Visual Supervisor

TO THE EDITOR:

How terrific it is to see Fred getting recognition for something only he (and very few others) can do.

Thank you.

DAVID RUBINSON
San Francisco, CA

SPECIFY EXCELLENCE!

from the company who pioneered equalization

• ACTIVE AND PASSIVE EQUALIZERS

18 different Models to choose from

• REAL TIME ANALYZERS

Octave Band, one-third and one-sixth octave

• BI-AMP AND TRI-AMP CROSSOVERS

Low-level at any frequency and slope

• NARROW BANDWIDTH NOTCH FILTERS

Control of room feedback and ring modes

• CUSTOM FILTERS FOR AUDIO APPLICATIONS

High-pass low-pass band-pass notch

SEND FOR OUR COMPLETE PRODUCT CATALOG

White

INSTRUMENTS, INCORPORATED
P.O. BOX 698
AUSTIN, TX 78767
(512) 892-0752

TO THE EDITOR:

Three cheers for Fred Catero (Getting Down to Two Tracks Fast and Other Stories, December 1980) via Ralph Hodges. Catero is an old-fashioned engineer, and in this case, being old-fashioned is also being new-fashioned. I heartily agree with his philosophy of recording and wish I had written the article first—although I'm not sure I could have expressed it as well as he did. I'm presently involved with a swing-style

Copies of db

Copies of all issues of db—The Sound Engineering Magazine starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to:

University Microfilm, Inc.
300 North Zeeb Road
Ann Arbor, Michigan 48106

Circle 18 on Reader Service Card

Introducing the Crown PZM, the second major advance in microphones in 100 years.

In 1876, Bell invented the first microphone.

Crown now announces the second microphone – the PZM™.

During the last century, microphones have been much improved, but they still employ Bell's basic concept: a movable diaphragm connected to a transducer, the whole assembly intended to be stuck out in the air somewhere near the sound source.

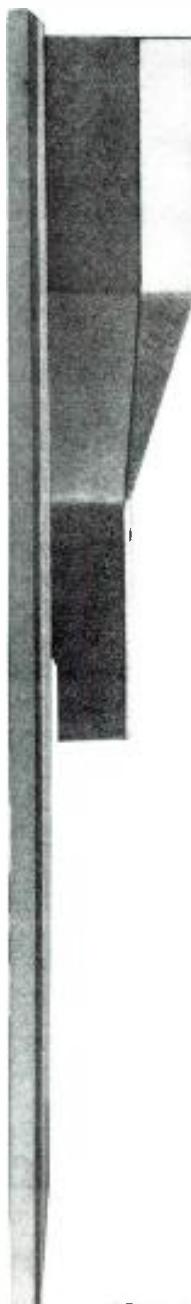
Comb filtering is a side effect of that design that cannot be eliminated.

Every Bell-design microphone demonstrates frequency response anomalies because of an inability to satisfactorily combine direct and reflected signals. Phase-induced amplitude cancellation and reinforcement are the inevitable result.

Crown PZM microphones eliminate comb filtering from the primary boundary because they detect sound according to a new principle, the Pressure Recording Process™. As a sound wave approaches a boundary (wall, table, floor) a pressure field four or five millimeters deep forms at the boundary, within which the direct signal and its reflection from the boundary add coherently and remain in phase.

The Crown PZM™ places a small pressure transducer into the primary boundary pressure zone, eliminating the possibility of phase-induced interference. The PZM concept thus provides a significant improvement in signal quality. Its small profile also improves microphone aesthetics.

The PZM pickup pattern is hemispheric, with no "off-axis" position.



Singers and speakers can move more freely around the PZM. Gain related to distance will change, but not tonal quality.

The PZM responds accurately to SPL up to 150dB. You can put it right inside a drum, a bass fiddle, or a piano. The PZM hears whispered conversations in an ordinary room at thirty feet. In certain situations where undesired ambient noise can't be eliminated, or in halls with poor acoustics, the PZM probably should not be used – it will pick up everything.

Singers, orchestra conductors, pianists, percussionists, broadcasters have all tried – and praised – the PZM.

Recording engineers find that the PZM suggests new mixing techniques. For small groups it now seems that the best place for a PZM is on the floor! Recording and reinforcement may well require fewer PZM mikes.

Several PZM models are now available, including a clip-on and recessed model for permanent installation.

The PZM is changing ideas about how a microphone ought to sound, look and be used. Find out for yourself how it might improve your own recording or reinforcement systems.

*Write for information on the PZM.
Or call us at 219/294-5571.*



© Crown International, 1980

CROWN INTERNATIONAL
1718 W. Mishawaka Road, Elkhart, IN 46517
(Mail this coupon to Crown.) Send PZM info. Hurry!

db 3

Name
Company
Address
City State Zip
Phone



1718 W. Mishawaka Road, Elkhart, Indiana 46517
Innovation. High Technology. American. That's Crown.
PZM, PZMicrophone and Pressure Zone Microphone are trademarks of Crown International

Circle 13 on Reader Service Card

www.americanradiohistory.com

PRO AUDIO SEATTLE

WE'VE GOT
IT ALL
UNDER ONE
UMBRELLA



Recorders



Consoles



SMPTE



Automation

And over 55 lines including:
AKG, Ampex, Annis, Auratec, Beyer, BGW, DBX, Deltalab, ElectroVoice, Eventide, Gauss, Ivie, JBL, Klipsch, Koss, Leader, Lexicon, Master Room, MRL, Neumann, Orban, Otari, Revox, Roland, Sequential Circuits, Scotch, Sennheiser, Shure, Sony, Sound Workshop, Stanton, STL, Tangent, Tapco, Tascam, Teac, Technics, UREI, Vega

PROAUDIO SEATTLE
Professional Audio Equipment and Services
(206) 367-6800
11057 8th NE, Seattle, WA 98125

Circle 43 on Reader Service Card

big band whose fourth album I did not record, but happened to be present at the recording session. The band members, who are friends of mine, kept coming to me and asking why everything sounds like "it's being recorded in a box?" It was not my job to be engineer or producer on that session, so I quietly told them I would explain later. After the session I explained that the mixing techniques used by the engineer for the session did not permit pickup of the studio ambience. In addition, the studio itself was, to use Catero's words, one of those "semi-anechoic sound-sink(s) that pass for a studio today." The band didn't have a chance. I will be recording their next album—elsewhere.

Also, can you tell us the album number of "Black Pearl" so we can check out Fred Catero's work?

BOB KATZ
Recording Engineer
New York City

db replies:

As of now there is no US distributor for the "Black Pearl" album. As soon as word of one reaches us, we will pass it on to you. If you would like to familiarize yourself with Fred's work, he was the engineer on all of Herbie Hancock's albums, and many of the earlier Santana albums.

db Calendar

APRIL

6-9 National Noise and Vibration Control Conference and Exhibition. Hyatt Regency O'Hare, Chicago, IL. For more information contact: NOISEXPO, 27101 East Oviatt Rd., Bay Village, OH 44140. Tel: (216) 835-0101.

12-15 National Association of Broadcasters. Convention Center, Las Vegas, Nevada. For more information contact: NAB 1771 N. St. Northwest, Washington, D.C. 20036. Tel: (202) 293-3570.

25 1981 Midwest Acoustics Conference. Hermann Hall, Illinois Institute of Technology, Chicago, IL.

MAY

5-7 National Sound and Communications Conference. Hyatt Regency and Atlanta Hilton, Atlanta, GA. For more information contact: Gerald M. Newman, 222 S. Riverside Plaza, Suite 1606, Chicago, IL 60606. Tel: (312) 648-1140.

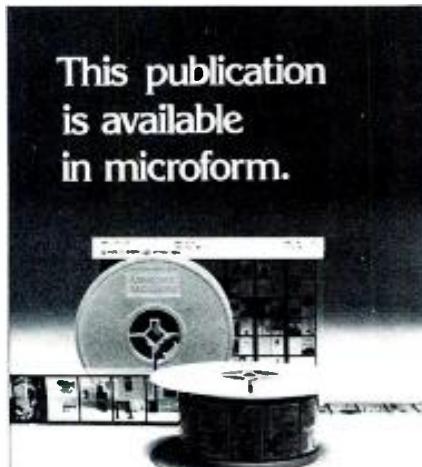
12-15 AES 69th Convention. Los Angeles Hilton, Los Angeles, CA. For more information contact: Audio Engineering Society, Inc., 60 E. 42nd St., Rm. 2520, New York, NY 10165. Tel: (212) 661-8528.

JUNE

10-12 APRS '81; 14th Exhibition of Professional Recording Equipment. Kensington Exhibition Centre, London, England. For more information contact: E. L. Masek, 23 Chestnut Ave., Chorleywood, Herts, U.K.

21-22 Audio Workshop. Concordia College, Moorhead, MN. For more information contact: Audio Workshop, Concordia College, Moorhead, MN 56560. Tel: (218) 299-4201.

This publication is available in microform.

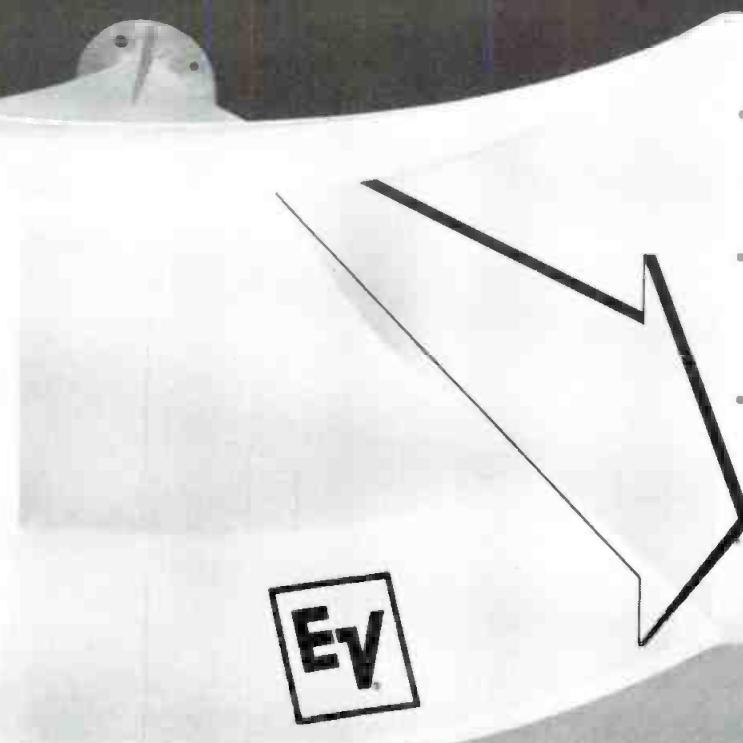
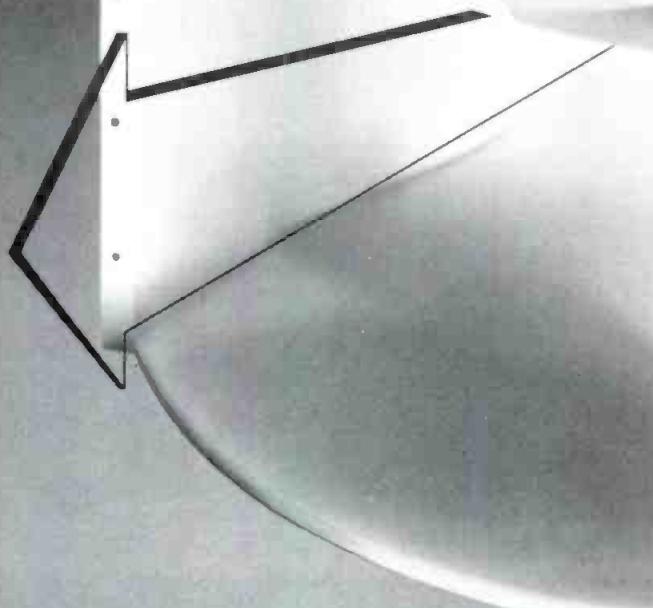


University Microfilms International

300 North Zeeb Road
Dept. P.R.
Ann Arbor, Mi. 48106
U.S.A.

30-32 Mortimer Street
Dept. P.R.
London WIN 7RA
England

constant directivity



The characteristic of a horn that directs all of the frequencies where you want them to go.

Most horns offer some control of the sound pattern they produce. The problem is that frequencies at the center of the pattern are different from those at the edges.

Unless you use HR Constant Directivity horns, that's the problem you'll have. To the audience this means unintelligible, too bright, too dull, and sometimes just plain bad sound at many seats.

These patented¹ HR Constant Directivity horns from Electro-Voice provide full-range frequency coverage and effectiveness of pattern control

unheard of before E-V engineers developed this unique design concept. Demand for the "white horns" has grown dramatically—almost completely by word-of-mouth. Once a sound engineer, musician or facility owner hears the difference HR Constant Directivity makes, a new demand is created.

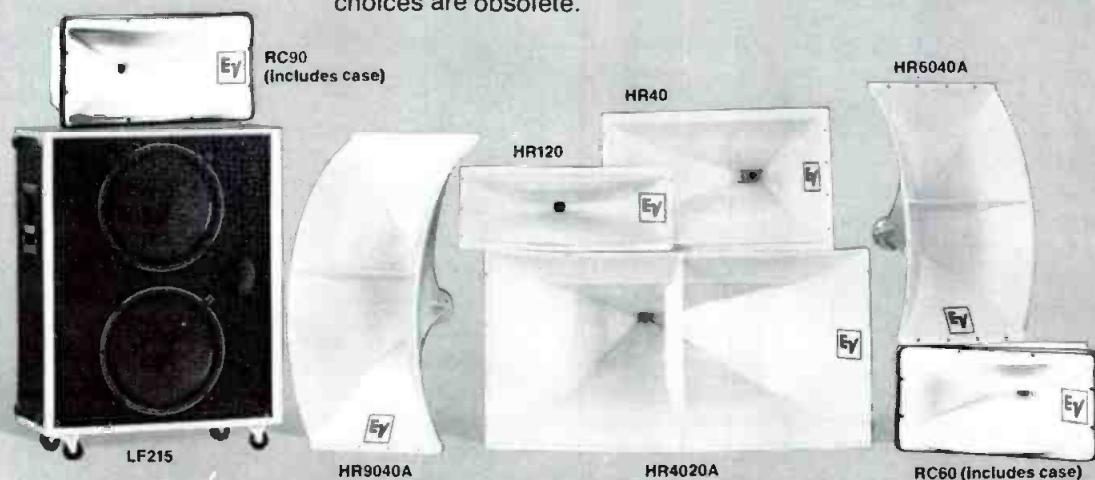
Ask someone who has used or heard them, or buy a pair and try them yourself. You'll probably hear that HR horns are so clearly superior that other choices are obsolete.

Write to Electro-Voice for more information. We'll send you a complete set of Engineering Data Sheets and a paper comparing the today performance of HR constant directivity horns with yesterday's promises. Include \$1 with your request, and we will put you on the mailing list for the E-V "PA Bible," a down-to-earth series of papers on the selection and application of professional PA products and concepts!

¹U.S. Patent Number 4071112

Electro-Voice
a Gulton company

600 Cecil Street, Buchanan, Michigan 49107



Circle 28 on Reader Service Card

STANDARD TAPE MANUAL



This valuable data book is for the AUDIO recordist, engineer or designer. Offered at \$45.00 you may order direct from publisher.

MAGNETIC REPRODUCER CALIBRATOR



This is induction loop equipment of laboratory quality for primary standardization of tape recorders and tapes. Send for detailed information, prices and formats.

R. K. MORRISON ILLUSTRATIVE MATERIALS

819 Coventry Road
Kensington, CA 94707

Circle 23 on Reader Service Card

FOR AUDIO
COMPONENTS
& LIVE PERFORMANCE

★ SH-1
STEREO HEADPHONE BOX

★ SHB-1
COMPANION
TO SH-1

HOLDS UP TO
SIX HEADPHONES

★ SC-1
STEREO COMBINER

SEND FOR YOUR FREE COPY OF OUR CATALOG

SESCOM, INC. (702) 384-0993
Professional Sound Division (800) 634-3457
1111 Las Vegas Blvd. North TWX 910-397-6996
Las Vegas, NV 89101 U.S.A.

Circle 19 on Reader Service Card

BARRY BLESSER

db Digital Audio

Distortion

We have talked about the nature of the noise inherent in the quantization process but we have not mentioned anything about distortion. In one sense, distortion is similar to noise, since both are differences between the input signal and its digital representation. This equivalence does not exist in the analog domain. Analog equipment has different mechanisms for noise and distortion. Noise comes from unwanted signals such as thermal effects, hum, crosstalk, etc. Distortion is the result of a non-linear relationship between input and output.

DIGITAL DISTORTION

We can draw an analogy by asking about non-linearity in the A/D or D/A transfer function. One could have quantization levels that create a gradual non-linearity. However, current specifications of DACs are in terms of $\frac{1}{2}$ LSB deviation between the actual level and the ideal level. The maximum value of the non-linearity, regardless of its shape, is thus limited to this value. So we can conclude that classic harmonic distortion will be on the same order of magnitude as quantization noise. Since these noise values are on the order of 60 to 90 dB, harmonic distortion will be on the order of 0.1 to 0.003 percent, referred to a full-level signal.

With such performance, the issue of harmonic distortion would appear to be irrelevant. Moreover, harmonic distortion can only come from a systematic non-linearity. For example, various-order components of harmonic distortion come from a non-linear transfer curve which contains small quadrature components, as seen in the formula:

$$E_{out} = E_{in} + A(E_{in})^2 + B(E_{in})^3$$

Here, the values of A and B determine the amount of second- and third-harmonic distortion. To see coherent second- and third-harmonic distortion in a digital audio system, the DACs would need to have a similar relationship. In general, there is no such systematic error, since digital errors are either

random or concentrated at certain points, such as the major carry. Being random, the errors are described as noise, not distortion.

We now reach an interesting conclusion: non-linearities and noise are essentially the same, except that the former is systematic, and the latter is random. Currently-manufactured DACs, used in both the A/D and D/A process, have an identical specification for these. There is, however, another manufacturing technique which produces DACs with a large amount of systematic non-linearity, and a small amount of random error. These are called segment or piecewise linear DACs. They are not widely used, and the typical digital audio system is unlikely to have such devices.

LOW-LEVEL DISTORTION

One interesting case of distortion is for low-level signals, as we discussed in a previous column. With a small enough input signal, a sine wave is represented by a square wave. This is because there are only two quantization levels to represent the input. In this case, the quantization error is really a distortion, not a noise. Since this may only happen for signals that are 90 dB below clipping, we usually ignore such distortions. Moreover, the inherent noise in the original analog signal means that such a small signal cannot exist. Hence, more than two quantization levels are spanned.

Most of the interesting forms of distortion in a digital audio system are not from the quantization process. Rather they are from the support process, such as the sample/hold, filters, and other defects. We will explore these next.

HIGH-LEVEL DISTORTION

When the input signal becomes large, there is a tendency for certain stages to begin to show special forms of distortion. We should remember that these distortions will appear to be large when we compare them to the 0.003 percent of the DAC, but they may be small compared to typical audio equipment.

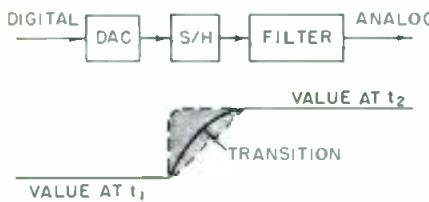


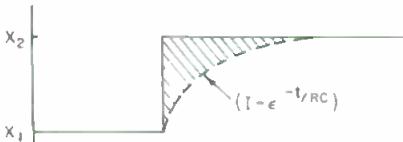
Figure 1. D/A conversion, showing various transitions between t_1 and t_2

Let us consider the sample/hold used at both the input and output. Each must make a transition from the previous value to the next value in a very short amount of time. In a system with a 50 kHz sampling rate, the input sample/hold may only have 5 μ sec to make the transition from -10 volts to +10 volts, since the other 15 μ sec are used for the conversion. This gives a slew rate of 4 volts μ sec. Although the amplifier may be able to move this fast, or even faster, the actual shape of the transition may be slightly non-linear. Instead of achieving the next value, there may be an error. The question of settling time is very complex and we will defer it. Just note that the slew rate must be larger by the ratio of sample time to sample rate. In the above example, the distance that the audio signal would normally go in 20 μ sec must be covered in 5 μ sec.

The distortion in the output sample/hold is much more interesting than that of the input. Notice in FIGURE 1 that the sample/hold is continuously connected to the output filter. With the input sample/hold, only the final value is used by the A/D. Certain distortions in the process of getting to the final value are meaningless if the final value is actually achieved correctly. With the output structure, not only must the final value be achieved, but the way in which the signal goes to this value is important. The graph in the figure shows an initial value at t_1 and a final value at t_2 , with a variety of transition shapes. The full signal is filtered and differences in the transition will be filtered also.

The ideal transition is the square jump in which the signal goes instantly from one value to the next, but no device

Figure 2. The ideal transition, compared to an RC exponential curve.



will actually do this. Another type of transition is an RC exponential; a non-linear transition. We should note that, in terms of what comes out of the filter, a straight line becomes non-linear, and an RC is actually linear. To illustrate this, we will go through the following arguments.

We take the ideal perfect jump as having no distortion and we assume that this produces the reference signal which we will use to compare other cases. Next, we compare this to the RC approach as shown in FIGURE 2. Since the filter is a low-pass, we may argue that the area of the difference is relevant; not the actual shape. A low-pass filter is an averager.

The shaded area can be shown to be proportional to $(X_2 - X_1)^2$. Notice that all of the above operations: delay, subtraction, and scaling, are all linear operations. The subtraction is like a difference or differentiation, which is a filtering operation. The net effect can be shown to be a slight error in the frequency response, not a distortion.

FIGURE 3 shows the case with a linear transition. Notice that the area of the shaded region is proportional to $(X_2 - X_1)^2$ since both the height and width of the triangle are proportional to the difference in signal values. This figure shows that changes in the difference between X_2 and X_1 produce large changes in the



How a Tweed Audio custom-designed console helped Grampian Television to make music and news!

For years, Grampian Television (Aberdeen, Scotland) had the problem of limited production flexibility. There were two possible solutions. The first was to modify a stock console (too expensive). The smarter alternative was a console from Tweed, custom-designed for newscasts and music programming, and built after in-depth consultation.

The module-based design includes 24 input channels with stereo or mono program out, and several cleanfeed outputs (international sound, mix-minus etc.) for linkup with other T.V. stations. A comprehensive system is built-in for communications between

studios, mobile remotes and camera crews, and in spite of the restricted dimensions, the console even has room for adding facilities if ever needed.

So, while the picture above looks like just another hardware photo, it is really a demonstration of how Tweed Audio solves a client's problems.

If you are looking for state-of-the-art answers to audio recording/mixing problems in a broadcast, film-production or recording studio, please contact us. We will be happy to supply anything from general specifications for existing consoles to "clean-sheet" designs to solve unique problems.

TWEED AUDIO

Los Angeles, Vancouver, Great Britain

Pinnaclehill Ind. Est.
Kelso, Roxburghshire
Scotland
Telephone: (05732) 2983
Telex: 7277633 Tweed G

Tweed Audio USA Inc.
12 Ilex Dr.
Newbury Park, Ca. 91320
Telephone: (805) 499-4764

COMMUNICATOR FEATURES

- Compatible with many hard wire PL systems including RCA, RTS, Clear Com, etc.
- True Hands Off Operation
- On-Hook idle usable range typical
- Push-Talk system ready to go
- Data interface optional
- Communicable with many popular headsets including PLANTRONICS M550, BEYER DT109, SHURE SM12, RODENWELL, ASTROLITE, MINILITE, TELEK, etc.
- Dimensions and Weight: 7.5 L x 4.7 W x 1.9 D 2 lbs
- Fast charge: SNAP-IN, NiCAD battery Pack
- A MARK 200 System consists of two MARK 200 Transceivers, antennas, batteries and carrying case, headsets

Swintek
ENTERPRISES INC
1180 ASTER AVENUE
SUNNYVALE, CA 94086
(408) 249-5598

Circle 39 on Reader Service Card

area. Thus, this effect is non-linear and it will produce distortion. The effect is third harmonic, since the shaded area will either add or subtract, depending on the sign of the difference.

We should comment that this error is largest for large differences between neighboring values. Hence, high-frequency signals produce the most noticeable slew distortion as seen from the output. A 20 Hz signal does not change value significantly in a 20 μ sec sampling interval, but a 20 kHz signal goes almost from full-positive to full-negative.

In the process of describing the effect, we have used exaggerated figures. Very small regions of slew-like distortion can produce degradation which is larger than that produced by the DACs quantization. The only forgiving aspect of this mechanism is that it shows up with the least likely signals: high level and high frequency. Most high quality digital audio systems will generally show some of this effect, but one requires a good spectrum analyzer to see it.

If you are keeping your wits about you, you may be asking: If the effect produces third harmonic distortion, and if we use high frequencies, shouldn't the harmonic component be filtered out by the filter? A test frequency of 18 kHz in a 50 kHz sampling system will produce a component at 3×18 kHz or 54 kHz. This is clearly outside the passband of the

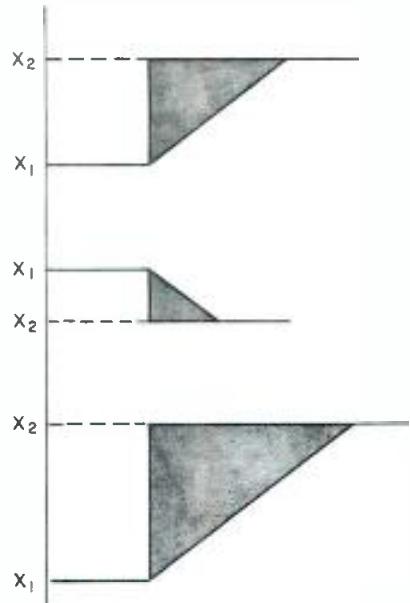


Figure 3. With a linear transition, the shaded area depends on the values of X_1 and X_2

output filter. The answer is difficult to demonstrate, but the third harmonic will beat with the sampling frequency to produce a 4 kHz distortion since 54 is 4 kHz above 50 kHz. A 17 kHz sinewave will produce a 1 kHz distortion. The slew distortion is not harmonic but it becomes a low-frequency component. This would not be true for an 8 kHz input, since the third harmonic is 24 kHz which would be filtered out. Also, a 4 kHz input would produce a 12 kHz third harmonic, which would not be filtered nor would it beat with the sampling rate.

In terms of perception, 16.5 kHz would be very bad since it produces a 500 Hz distortion. The input signal is not audible but the distortion is clearly heard. No signal masking will cover the 500 Hz.

TESTING

It should now be clear that the most difficult test for a digital audio system is a full-level signal at approximately one-third the sampling frequency. If the system performs well in this case, there is probably no distortion problem in the entire chain. It is a simple test and it does not require fancy equipment if you use your ears. Normally, one could not use an audible test since the distortion component was a multiple of the fundamental. The ear, as well as all the support monitoring equipment, generates more third harmonic than the equipment. However, in this special case, the distortion is moved to a new frequency which is very audible. None of the other equipment in the studio can move a single harmonic to a new frequency. Intermodulation tests are similar but they use two input signals to create the beats. In this case, the sampling frequency replaces one of the reference signals.

NO OVERLOAD. NO MATTER WHERE YOU USE IT.

In-studio or on-location from a whisper to the roar of a full-blown rock concert, nothing can overload our MD 421. *Nothing.*

SENNHEISER
ELECTRONIC CORPORATION
10 West 37th Street, New York, NY 10018
(212) 239-0190
Manufacturing Plant: Bissendorf/Hannover, West Germany
© 1980, Sennheiser Electronic Corporation (N.Y.)

Circle 41 on Reader Service Card

JBL's new 7510.

The automatic mic mixer that thinks before it speaks.



JBL's new 7510 advanced digital/analog mic mixer actually thinks before it speaks by activating or deactivating up to 24 input channels. Automatically. Now you don't need an operator to constantly adjust controls. The 7510 does it all. Quickly, precisely and without error.

Versatile modular design makes the 7510 an all-around performer. In churches, meetings, court-rooms and council chambers. Delivered with a single, 4-channel input module, the system is expandable to 24 channels. And each panel features controls for level, threshold, release time and mode selection.

The 7510 offers a variety of other advantages over conventional mic mixers. Automatic mic on/off

and output level correction permit hefty power gain without the "howling" of feedback. Turn-on features a zero crossing detector. So there are no pops or clicks!

Advanced level sensing circuits trigger extremely fast attack — 30-60 nanoseconds. This quick rise makes it ideal for gated mixing. The 7510 offers separate outputs for every input. So the user can program to match each need. When in the priority mode, all inputs in automatic are muted by selected lead mics.

At the same time, a unique digital attenuator decreases the system's output gain by 3 dBs each time the number of live mics doubles. The feedback stability margin remains constant regardless of the number of mics

operating. And the system's threshold circuits can distinguish between program signals on one mic and ambient noise on all mics to within 1 dB.

Other features include 48-volt phantom power supply, Master VU meter. And the system fits easily in three EIA rack spaces.

The 7510's low distortion, low noise, flat response and wide input dynamic range make it perfect for all sound reinforcement applications. It's also ideal for noise gating in recording studios and for broadcast and live music reinforcement applications.

JBL's 7510 delivers what it promises.

James B. Lansing Sound, Inc., 8500 Balboa Blvd., Northridge, CA 91329

Frequency Response	
Input Module	20 Hz-20 kHz ± 0.5 dB
Output Module	20 Hz-20 kHz ± 0.5 dB
Overall System	20 Hz-20 kHz ± 0.5 dB

Maximum Gain	—
Overall System	— 80 dB

Main Output Characteristics	
Maximum Output	· 23 dBm
Load Impedance	600 Ω or higher loads

Total Harmonic Distortion	
Mic In to Direct Out	0.2% maximum 35 Hz-20 kHz
Mic In to Main Out	0.2% maximum 35 Hz-20 kHz

Automatic Mic Functions	
Input Rise Time	30-60 nS
Input Release Time	100 mS to 5 S, adjustable

JBL First with the pros.



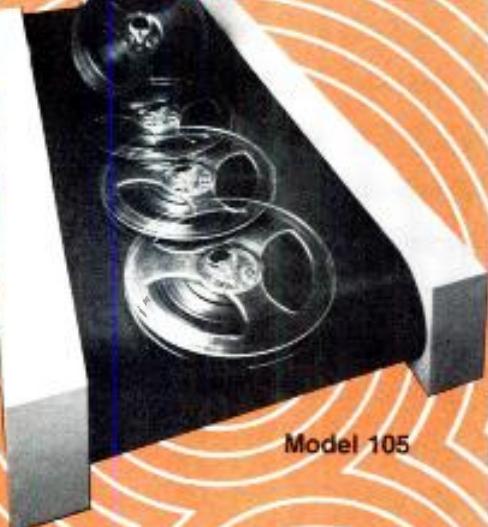
Circle 37 on Reader Service Card



NORMAN H. CROWHURST

db Theory & Practice

Now you
hear it...



Model 105

Now you don't.

The Garner Audio Tape De-gaussers are truly the most flexible and thorough audio tape erasers available. With them, you can erase cassettes, reels, and cartridges in just four seconds. And, we guarantee that your erasure will meet the most stringent recording standards. It really is a case of "Now you hear it... Now you don't."



GARNER INDUSTRIES, INC.

4200 No. 48th St., Lincoln, Nebraska 68504
Phone (402) 464-5911

**There's more
to hear...**

Just write Garner Industries for
more information:

NAME _____

TITLE/POSITION _____

COMPANY _____

ADDRESS _____

CITY _____ STATE _____ ZIP _____

Circle 32 on Reader Service Card

- The principle we want to explore here, involving both theory and practice in problem solving, can appear in a variety of contexts. Our first awareness of it came in an engineering context, when designing equipment. You would start with theory and run through a series of designs, based on prior data, until you arrived at one that, in theory, was close to the requirements you had established from the parameters of the problem.

Then you went down to the model shop, and got the design made up for testing. More likely than not, the finished product would differ from its design spec, and you would have to adjust the design figures and try again. Eventually you'd get a sample that met the specs, and you'd put it in the system for which it was designed, only to find it didn't work for some reason you had failed to anticipate.

Solving a problem in that way can become a very protracted proposition. What bothered me, as a young engineer, was its indirectness; it was like shooting in the dark. We wanted more direct methods.

Back in my days as Chief Engineer at Tannoy, we developed those more-direct methods and, as a result, we were ahead of our competition for several years simply because we were solving problems while our competitors were trying to write down what the problem was. A reputation for that kind of capability gets around.

Often, the direct method involves

different approaches to the same basic problem. Back in those days, the only kind of signal generator was one that used frequency-selective networks with variable feedback. There were variations within that theme, such as beat-frequency generators, in which two ultrasonic frequencies were beat together to get the audible one we wanted, or phase-shift networks that directly selected the wanted frequency. But they had one common weakness. All the reactive networks carefully selected the frequency at which the generator ought to oscillate. Then, the feedback determined whether an oscillation started and built up to a stable amplitude, or whether it died away and quit oscillating. While some excellent generators were made that way, they were always difficult to adjust, and could easily go wrong.

Eventually, someone concluded that we were going the wrong way—or not the best—way about it, and the function generator was born. This uses a whole new approach, which is amazingly more direct. We've covered that before, so we'll not belabor it here.

All of this connects into music synthesizer technology, which didn't get its start until many years later, but which applies the same principle, and carries it further. In those days, the nearest thing to a synthesizer was an electronic organ. And organs of those days used individual generators of the same type; either one for every note on the organ, or one for each of the notes in the top octave, with

The Tascam 16-Track System.



Chair not included.

**All this for
the price
of a 2-inch
recorder.**

If you don't need 2-inch compatibility or 24 tracks, you don't have to pay the price for expensive 2-inch hardware.

Instead, you can own the heart of a 16-track studio.

The Tascam Model 15 Mixer. 24-in 8-out 16-track monitor with a comprehensive cue system that can be fed simultaneously by 48 signals.

The Tascam 85-16 Recorder/Reproducer. A 1-inch transport with 16-track integral dbx.*

And the Tascam 35-2B Mastering Recorder. A 1/4-inch 2-track with integral dbx.

With this Tascam 16-track system, you'll get professional sound at an affordable price. And because it's all new equipment, you save even more with tax credit potentials.

Write to us today for the name of your nearest authorized Tascam dealer. He'll show you how a Tascam 16-track system can fit your studio needs and your budget.

TASCAM

TEAC Production Products Group

©1980 TEAC Corporation of America, 7733 Telegraph Road, Montebello, CA 90640

Circle 14 on Reader Service Card

*dbx is a registered trademark of dbx inc.

dividers for the lower octaves.

But digital technology, which is related to function-generator technology and is also far more direct in basic concept (if more complicated in the details of its implementation), found a way past that. In those days, if you wanted to tune an electronic organ, you had to adjust the frequency of every note in the first type, or of every note in the top octave in the second.

Digital technology made it possible to use a single, ultra-sonic master generator, and to derive all the notes used from that by division. That was a more sophisticated way of applying the divider

principle that had formerly derived lower octaves from the top one, by a succession of divisions-by-2. But now division by any preselected set of numbers was possible by use of a digital programming technique, so that one master frequency could generate all the notes on the musical scale.

If the master frequency was shifted, every note on the instrument shifted to correspond. Thus a whole organ, or synthesizer, can have its pitch shifted with one simple adjustment. Again, a more direct approach to doing the job.

Back to engineering in the lab. You have a piece of experimental equipment

in front of you, made to your design, but it doesn't work in the way you predicted. What do you do? Basically, there are two approaches. One is to check a lot of things to find out what is not happening that you predicted should, then apply theory to see why it might not be happening, "zero in" on the problem, and finally make changes to overcome it.

To many, that seems like a long process. They want to get a quicker answer. So they start "trying things." They remember other instances where strapping terminals together, making additional connections, or changing values, solved such a problem. So they start making such changes, willy nilly. If they could just "hit on the right one," it would be quicker than the process we first described.

There are two things wrong with that. First is the fact that usually it becomes a case of "lazy people taking the most pains"; what seemed quicker turns out to take much longer in practice than in theory, and probably never produces a solution. Secondly, even supposing you find something that "works" this way, you don't know why it does. Usually it turns out that you've solved one problem, and made two more that may not show up until later, perhaps when a few million of the products are out in the field!

Having looked at the general nature of such problems and the approach to their solution in an objective or engineering context, let us turn to a subjective context, in which our engineering technology may find a way to help. What we are really talking about is a more-intelligent relationship between man and machine. Too often, we try to reverse their roles. People think creatively, machines don't. But often people who can't (or don't want to) think, expect machines to do the thinking for them.

Many readers know Don and Carolyn Davis, who put on the Syn-Aud-Con seminars. I remember talking to Don one day, quite a long time ago. He had just acquired a Hewlett-Packard calculator for something like \$6,000.

Don confessed that in high school and college he was a real dummy in math. It just didn't make sense to him; but since acquiring his calculator, he had sat at his desk "doodling" with it, and came to the conclusion that he was not really a math-dummy after all! He just hadn't been taught right. So, he re-discovered math all by himself, sitting at the calculator keyboard.

He was so taken with the value of this approach (who wouldn't be?) that he wanted to pass it on. He went to schools and tried to persuade them to invest in such an instrument, with little or no success. The problem? Teachers are afraid to venture into unchartered waters, and are not about to turn children loose on such a thing—as Don had done for himself—because to do so would mean a loss of "control."

Since 1941, audio professionals have been discovering that AB Pearl Mikrofon Laboratorium of Sweden make some of the finest quality microphones in the world.

Under our new name we're proud to introduce the same old thing: precision craftsmanship, superlative performance and value. Our name

The Swedish Steal Has A New Name: MILAB has changed but our dedication to provide the audio pro with a superior alternative has not.

Whether it's the DC-63 variable pattern dual-membrane condenser, the DC-96 cardioid condenser, VM Series condensers, F-69 dynamic, CL-4 electret or an XY stereo mic, your professional audio dealer

should be contacted to give you a chance to audition for yourself the performance and value of the Swedish alternative:

MILAB.

DC-96

MILAB™

We're Still The Swedish Steal!

For product information write or call your nearest MILAB distributor:

Worldwide Marketing—Creative Trade CTAB AB, Knutsgatan 6, S-265 00 Astorp, Sweden. Tel: 42/515 21

United States—Cara International Ltd., P.O. Box 9339, Marina del Rey, California 90291. Tel: (213) 821-7898

United Kingdom—Future Film Developments, P.O. Box 4BS, London W1R. Tel: (01) 437-1892

Australia—Werner Electronic Ind. Pty. Ltd., P.O. Box 98, Kilkenny, S.A. 5009, Tel: (08) 268-2766

© 1980 MILAB

**Introducing the first line of
professional recorder care products.
The Proformance Series.™
Designed and created exclusively
for you, the professional. Who else
but Nortronics could deliver
products like these.**

Nortronics, the undisputed world leader in magnetic heads. We make the heads others are compared with. Over 25 years of engineering skill and expertise are behind every head. Today, high quality and

technical innovation have become Nortronics' trademarks.

Now get that same expertise in recorder care products. The Proformance Series. We first pinpointed the industry's precise requirements. Designed to meet them. Established critical tolerances—tolerances that meet or exceed any requirement you may ever have. The key to everything is simplicity. In short, each product will exceed your highest

expectations. Because that's the way we do things—professionally.

Ask your distributor about the
Proformance Series. Or call
1-800-328-5640 for the distributor
nearest you. In Minnesota, call
612-545-0401.

PROFORMANCE SERIES™

BY NORTRONICS.

8101 Tenth Avenue North
Minneapolis, MN 55427 U.S.A.

**Keep them as great as
we've built them.**



Circle 26 on Reader Service Card

Today, I encounter a different problem. For about \$9.95, any youngster can acquire a calculator with the four basic functions on it. That's a far cry from Don's \$6,000 investment, and so it has become impossible to keep them out of the classroom. Most teachers have given up. Now, it's the parents who ask about the practice of allowing kids to use calculators. They feel that—unless the kids have to memorize the multiplication tables and other things that were required years ago—they're just not going to "learn."

I can't quite agree, and I'm sure Don Davis wouldn't either. If the calculator could really be used as a "cheater," then I'd have to agree. But just as the much-more-expensive one in which Don had invested was a useful tool in learning mathematics, so too can these cheap pocket ones be. It's not *whether* they are used, so much as *how* they are used.

Back in the days when the Federal Communications Commission had just approved the stereo multiplex system that is in common use today, a publisher asked me to get a book ready that would scoop the market on this subject. There were already about three ways to design a multiplex decoder, and the technical publications were full of ads from all the leading tuner-makers. "Ours uses the best system" campaigns gave the impression that any day now, you'd be able to

go to your corner radio store and buy one.

Thinking this was so, I wrote to all the manufacturers for information to include in my book, but got no answers. After a follow-up letter, I started getting phone calls from Los Angeles, Chicago, Philadelphia, Boston—all over. They all said the same thing: "Hop a plane, we'll pick up the tab."

It struck me as a rather expensive way to distribute information: why not just stick it in the mail? I soon found out. Arriving at my various destinations, I was met at the airport, treated royally, wined-and-dined by the executives, and then ushered into the lab, where one or more engineers were trying to make the prototype work. They'd read all the theory, decided on a design approach, but just couldn't make it work.

In each case, I'd spend a few hours in the lab, examine the problem, perhaps help remove a roadblock or two, and leave. I began to realize that the roles of man and machine were being reversed by our society.

In school, students are taught to solve problems according to specific type. The teacher must know how, so he or she can show the student how. Then, the student knows how. But anything that has not been shown, cannot be solved. This is a serious malady in today's world, and total reliance on computers just aggravates it. But the

same change that remedied things back then, still can.

People think that anything a computer spits out is infallible, because a computer "can't make mistakes." Maybe some are a bit disillusioned about that, particularly people who spend \$100 on their credit card, and get billed for \$100,000. That's apt to strain anyone's faith in infallibility. But they still don't ask, "Why do mistakes happen?"

The musician who plays a classic (non-electronic) instrument requires skill and training to master his instrument, doesn't he? Does a person who plays a synthesizer, or an electronic organ, not need such preparation? Perhaps this is a silly question, but I think you get my point. It may be a more sophisticated instrument than its classic forerunners, but still it can only do what the musician makes it do. (Maybe I'll get some arguments on that.)

Mistakes in using calculators or computers happen mainly because the operator hits the wrong button, or hits the buttons in the wrong sequence. Sometimes this happens because the operator doesn't really understand the machine (or maybe even the problem) and is just "trying things."

On an electronic music instrument, the musician/operator can waste a lot of time "trying things." Or, he can spend some time learning to understand the system's sophistication. ■



Our Stereo Synthesizer isn't just for old mono records.

Applications of the 245E are limited only by your imagination:

- save tracks by recording strings, horns or drums on a single track and spreading them in the mix
- create stereo depth from synthesizers, electronic string ensembles, and electric organ
- create a stereo echo return from a mono echo chamber or artificial reverb generator
- use one channel to create phasing effects

It's a dramatic, highly listenable sound that's fully mono compatible—just add the channels to get the original mono back. (If you get bored, you can always process old mono records into pseudo stereo.)

Your Orban dealer has all the details. Write us for his name and a brochure with the complete 245E story.

orban

Orban Associates Inc. 645 Bryant Street, San Francisco, CA 94107 (415) 957-1067

Circle 34 on Reader Service Card

The Evolution Of Sound.

An evolution of high-powered audio that develops pure, unadulterated sound. Uncolored and vibrant. The evolution that begins with TAD.

As evidenced by the TD-4001 compression driver. With pure Beryllium diaphragms and surrounds that are extremely lightweight and rigid, the TD-4001 provides unsurpassed high frequency response and sensitivity.

The TD-2001 compression driver combines the features of the TD-4001 and packs them into a driver that is half the size. Yet, the TD-2001 delivers the kind of high frequency sensitivity that outperforms drivers of any size. Without sacrificing power handling.

The TD-4001 and TD-2001 compression drivers. Their unprecedented clarity represents a milestone in the evolution of sound.

Then there's the TAD TL-1600 15" low frequency loudspeaker series that advances sound delivery another step. With select high purity materials and precise machining tolerances, the powerful Alnico Ring Magnets produce the kind of strong linear fields necessary for the most demanding application. High-power voice coils on four-inch heat resistant glass fiber bobbins assure long-term dependability. The TL-1600 series, a continuation in the evolution of sound.

Pure sound evolves with quality and performance in equipment.



The evolution of sound begins with TAD.

TAD Technical
Audio Devices

A division of U.S. Pioneer Electronics Corp.
142 Redneck Ave., Moonachie, N.J. 07074
(201) 440-8234 Telex: 133484 PIONERUSA MOON

Circle 31 on Reader Service Card

db Sound With Images

• Most of us think of our eyes as being our most important sense organs. It's true that we learn mostly by what we see, but it has also been found that loss of hearing has a more disastrous effect on the psychological development of humans than blindness. For all you audio people out there, this should be easy to "see." All your critical hi-fi listening might go in one ear and not the other.

It's true that in the early development of the human body, the ear was originally meant to be used for balance. That's why a damage or defect in the inner ear causes vertigo, car-, plane-, and boat-sickness. (Those wild rides at the amusement parks are made to stimulate this sense, and can either titillate or nauseate, depending on your inner ear.) But the ear also developed to hear, and although parts of it are not as useful to the human as they might be to animals, down deep inside (about an inch or so below the outer ear) beats a very sensitive mechanism, which,

through all its complex connections, allows the brain to receive those sensations we call sound. Names like ear drum, tympanic membrane, tympanic chamber, cochlea, ossicles, auditory meatus, eustachian tube, oval window, and round window identify a few of the parts that make up the mechanism that allows us to hear.

Although we don't realize it (or take it for granted), the entire mechanism is really a marvelous device. While the part of the ear outside the head doesn't work well for us (dogs, for example, can literally perk up their ears to help gather the sound they want to hear while humans have to cup their hands and pull the outer ear away from the head to do the same thing), we can concentrate on one sound even though it may be buried among many others that are higher in level. You might be interested in knowing that the small air space leading from the outer ear to the ear drum has a resonant

frequency of about 4,000 Hz. It might also be interesting to note that the ear drum is extremely efficient in that it reflects very little of the incident sound and transmits almost all of it to the area and parts behind it. That part of the hearing mechanism acts like a preamp. Pressure on the ear drum is only about one fifth the pressure passed on to the inner connections which eventually feed through the eighth cranial nerve (there are 12 of these nerves in total) to the lower brain.

With this kind of mechanical advantage, it's quite obvious that the loudest sounds (mostly man-made) could cause damage to the hearing mechanism, so the ear was equipped with two striated muscles (the smallest in the entire body) to try to increase the system's internal friction and prevent the 12,000 little hairs (that's all we have) in the system from being bent out of shape. Sometimes though, the incoming wave front is so strong and so sudden, that the muscles, which need about 60-150 milliseconds to contract, don't react quickly enough and there is some damage.

Consideration of some other small details will further illustrate how marvelous the ear is. For example, the area of the ear drum is about 70 sq. mm. The oval window, through which vibrations are finally transmitted, is only about 3 sq. mm in area. In spite of these tiny dimensions, the approximate minimal energy level of a 1,000 Hz signal that we can sense is 0.0002 dyn/sq. cm. If our hearing were more sensitive, it is conceivable that we would be disturbed continuously by molecular bombardment which might allow us to hear temperature differences. (I wonder if that's how crickets and their kin can tell whether to rub their back legs faster or slower.)

It's also amazing that the human ear can distinguish about 250 distinct intensity levels, and about 1,000 pitch discriminations in the range from 20-16,000 Hz. (Some of us humans have what is known as perfect pitch and many are greatly stressed by screeching cars and subway trains, while others must cover their ears during the sound of rumbling trains or roaring planes.) The human ear can be sensitive to "volume" or "mass" of sound as distinguished from loudness (which phenomena are not easy to explain but for which we now have different meters and different nomenclature).

Take a Waters slim-line space-saver

New MM thin-line faders are great for small portable mixers or lighting consoles. Only half the width of conventional faders, our 3/4-inch wide MM series lets you build a lot of mixing or control capability into a limited space. No performance compromise, either. You get Waters' proprietary MystR[®] conductive plastic resistance elements. Glass-hard. Smooth. Long-lasting. You get Waters' computer-controlled curve shaping for highest tracking accuracy. And, you can choose 2 3/4-inch or 4-inch travel; 600 ohm or 10,000 ohm impedance; linear, audio, or true log characteristic. For complete information, circle the reader service card, or call Bob West at (617) 358-2777.

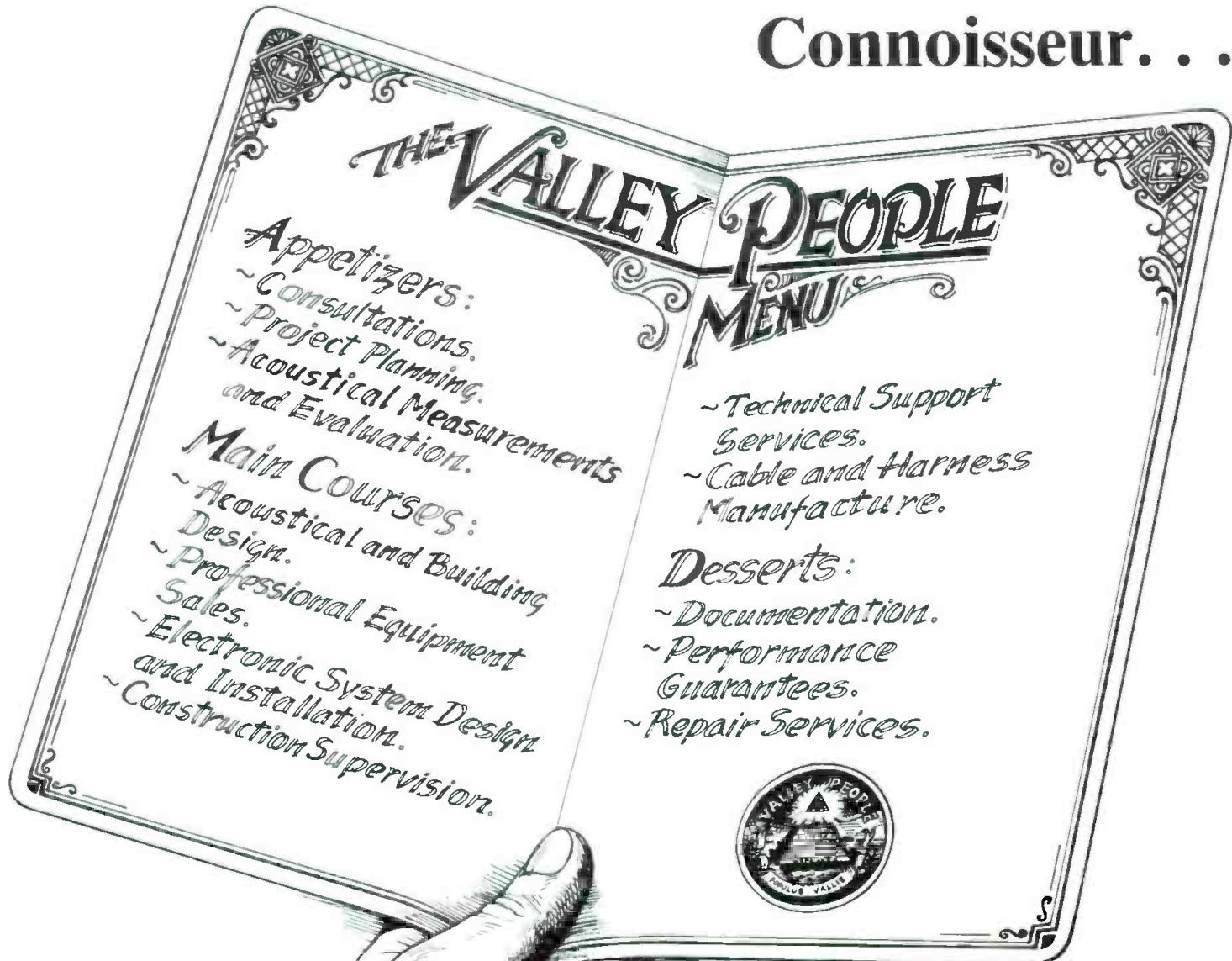


WATERS MANUFACTURING, INC.
LONGFELLOW CENTER, WAYLAND, MA 01778

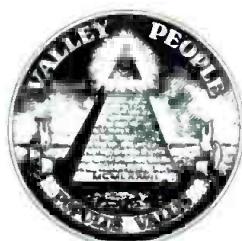
Circle 40 on Reader Service Card

For the Audio/Video Facility

Connoisseur . . .



Why settle for less than gourmet cuisine to satisfy your audio/video "hunger"? □ For many years, the Valley People "chefs" have been involved in the innovative design, planning, construction and renovation of major audio/video facilities around the world. This experience allows Valley People to offer you the most extensive consultation services available anywhere. □ Plus, the full line of equipment we sell comes from manufacturers who are as serious about quality as we are. □ Richard Lee and Bob Todrank are ready to serve you now. Give them a call to discuss your particular needs. Be sure to ask about our helpful *Facility Planning Guide*. □ As for price . . . well, we're no fast food franchise, but we guarantee satisfaction. Besides, the bitterness of poor quality lingers long after the sweetness of low price is forgotten.



VALLEY PEOPLE, INC.

P.O. Box 40306
2820 Erica Place
Nashville, Tenn. 37204
615-383-4737
TELEX 558610 VAL PEOPLE NAS
a merger of Allison Research & Valley Audio

Circle 36 on Reader Service Card

EASTMAN RECORDING INSTITUTES

Eastman School of Music
of the
University of Rochester

NEWLY DESIGNED AND EQUIPPED
PROFESSIONAL FACILITIES

Basic Recording Techniques
June 22-Aug. 1

Advanced Recording
Techniques July 13-Aug. 1

Instruction by leading professionals in the fields of audio recording and engineering. Covers microphones, studio setups, remotes, classical recording, signal processing, maintenance, etc. Labs, recording sessions, mixdowns. Ros Ritchie, director of Eastman Recording Services, coordinator. Credit and non-credit.

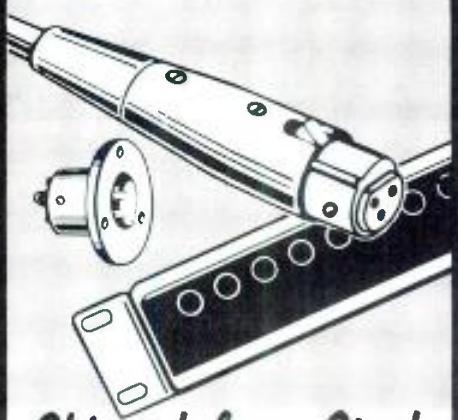
For information and applications, write: Summer Sessions, Dept. N, Eastman School of Music, 26 Gibbs St., Rochester, N.Y. 14604.

The Eastman School provides equal opportunity in admissions and student aid.

Circle 22 on Reader Service Card

Audio Connectors Jacks, Plugs Patch Cords

SWITCHCRAFT BRAND



Shipped from Stock

ASK FOR OUR CATALOG
OF PRO-AUDIO SPECIALTIES

Polyline
Corp. 312/298-5300

1233 Rand Rd. • Des Plaines, IL 60016

18

The ear has many more marvelous things about it which would take a good deal of time to illustrate, but the reason for all this is to show how much there is that can be different from one ear to the next. Some people need louder sound in order to get the full range they are listening for, while others are so sensitive that they turn high fidelity controls down for comfortable listening. Some of us raise the bass while others raise the highs during the playing of the same piece of music. Some like the deep richness of a "natural" bass tone, while others can't tell if the bass is real or artificially flavored. Some buy certain hi-fi components for whatever quality they are seeking, while others will think lo-fi systems in a single housing totally adequate. It's all a matter of individual taste, and that's why there are so many manufacturers and models of speakers and components available in the market place. There are also some individuals who think that price alone makes the difference, and will buy only the most expensive items for the home system even though they may not be able to distinguish between a violin note and a goose honk.

This leads to the next point. In recent columns in this corner, the question of sound during audio-visual presentations was raised. It seems only fair that the time, effort, and cost of the visual part of the show be matched by the audio part. It seems such a shame and waste to have so much put into the pictures and so little into the sound. A great visual presentation can be badly hurt by poor sound, and a fair visual show can be killed by bad sound. But what is good sound?

Shortly after several of these columns came out, inquiry was made by readers as to what speakers we would recommend for audio-visual shows. We are very reluctant to suggest any particular brand of speaker, or model. Not all the speakers available are acceptable, of course, but to suggest certain units in preference to others would be unrealistic and unfair. What we would like to do, though, is to suggest some of the parameters that should be used as guides for selection of speakers to go with shows.

For instance, think of the size of the room in which the show will take place. Each situation might be totally different from the next. If the room is very large, and the audience is expected to fill the room, more sound has to be distributed than if the room is small and will contain only a few people. It's always best, if possible, to have the loudspeakers at the front of the room near the screen for greater realism. Sometimes this is not possible, because a lectern with a live microphone is right in line with the loudspeaker and might result in feedback. Perhaps the speakers can be moved off to one side, farther away from the screen and in line with the lectern to

minimize the possibility of feedback. This might result in sharper angling of the speakers toward the audience. Be sure however that there is not a "hole" in the center near the front. If the stage platform is raised fairly high, perhaps the speakers can go under the screen. Just remember that there should not be too much sound for the first row in order to get a little sound into the last one. Perhaps the speakers can be raised above the first row and then angled to give enough sound for all. It all depends on the room and the setup.

Speakers should never be under-powered. If the room is large, a few simple calculations considering the distance to the last row and the level desired will give the requirement the speaker should meet to provide that sound. Like with all other sound equipment, never work a speaker at its top edge of operation. If you raise the volume, the speaker should not break up or crackle. Also, keep in mind the amplifier that is feeding the speaker. Don't run that unit at its top, either. Consideration must also be given to hard surfaces that can reflect sound. For example, putting speakers in a corner of two hard walls might possibly provide more bass than you want or need. Carpeting and drapes will absorb some of the highs and leave you with more boom than you want.

Speakers should also be efficient so that the amplifiers don't have to work so hard to get the volume out. If you find the speakers can handle them and need more power, use more powerful amps. There are many criteria to finding the right speakers, but they should all be full range, especially if you have a good music score to play. You can get away with smaller range if all you have is a voice track, but there are a lot of people out there who consider themselves audio or hi-fi experts because they have "good" systems at home, and they are all in your audience. Don't let them down. Provide the best sound possible, to all of them. One last thing to consider. Don't check your sound level in the back of the room without thinking of the people in the audience. This does not mean you have to wait for them to get seated before you set your levels. What it does mean is that they will absorb a great deal of sound when they are in their seats, and you should set your level during testing accordingly higher than with the house empty. You might think of having a member of your crew in the back with a walkie-talkie when the show starts and again during the presentation to keep tabs on level and let you know whether to raise, lower, or leave it alone. He (or she) can also check on the slides and films at the same time, so that you know you're doing your best to give the attendees the best audio to go along with the best visual presentation.



Technological Leadership. It's the Reason You Should Buy Ampex Audio.

Technological leadership. It's the reason you buy Ampex equipment. And it's the same reason you should buy Ampex audio products.

Our ATR series recorders, the ATR-100 and the ATR-700, for example, are ideal audio recorders for broadcasters. You'll discover impressive performance and dollar-saving reliability when you put these hard-working recorders to use in your broadcast operation.

THE ATR-100. THE PROFESSIONAL.

The ATR-100 is the standard of excellence in the audio recording industry. Its reputation for low distortion, low wow and flutter, and phase corrected equalization is unsurpassed.

For broadcasters, a cue amplifier and editing kit are available. Finding an edit point is as easy as turning the capstan knob.

THE ATR-700. A RUGGED PERFORMER THAT'S ALSO PORTABLE.

Our ATR-700 is a master performer. Perfect for those situations where quality is as important as efficient, trouble-free production. You'll find the ATR-700 to be a rugged performer for news and commercial assignments in the field, as well as a first class addition to your audio equipment in the studio.

You'll get standard features like important controls up front for easy operation, plug-in printed wiring assemblies for efficient service, and a built-in 4 in 2 out mixer.

Circle 15 on Reader Service Card

www.americanradiohistory.com

WHEN YOU THINK AUDIO, THINK AMPEX.

Ampex was there first. And we're still the leader. So, when you think audio for your broadcast operation, think Ampex audio.

Call your Ampex dealer today for the full Ampex audio story. In a competitive business, it pays to get the edge.

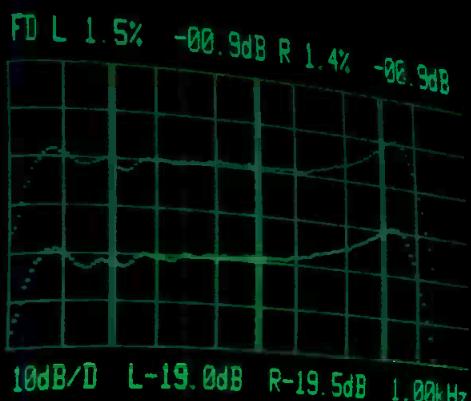
GET THE AMPEX EDGE.

AMPEX

Ampex Corporation
Audio-Video Systems Division
401 Broadway, Redwood City, CA
94063, 415/367-2011

•S

1500A TAPE RECORDER TEST SYSTEM
SOUND TECHNOLOGY



VERTICAL DISPLAY
DOWN
PANO
PUT
PUT REF
DISPLAY



POWER ON

CURSOR POSITION

LOW FREQ LIMIT SET

COPY

OUTPUT SW/SET MODE
MONITOR MANUAL SINGLE REPEAT

START OUTPUTS STOP INPUTS

INPUTS

THE INTELLIGENT
CLEANS UP

The Sound Technology 1500A

It's the first microprocessor controlled audio measurement test system. It can do in minutes what used to take hours with more conventional and ordinary test set-ups. And, it can show you things you've never seen before.

Designed around the most advanced microprocessor hardware, the 1500A will show you the whole story on an integral CRT with adjustable cursor. Push a "Copy" button, and it delivers a hard-copy printout from the optional VP-150 Video Printer.



What Will It Do?

Conceived to be the ultimate precision test instrument for tape recorder analysis, the 1500A evolved into a comprehensive audio test system for many applications. Here's just a small sample of the varied jobs it will do:

- Complete tape recorder mechanical and electronic performance checks
- Thorough phono cartridge analysis
- One-third octave spectral analysis
- Evaluation of audio quality for VTR's
- Acoustical room analysis including microphone and loudspeaker measurements



- Quality control for high speed tape duplication systems
- Semi-automated production testing
- Research and development for the audio tape manufacturer
- Quality assurance for the audio distribution network
- Exclusive asynchronous inputs and outputs for remote location testing (satellite, transmitter, studios, etc.)

Here's the kind of data you can get:

- Frequency Response
- Azimuth at 4 discrete frequencies
- 2nd and 3rd Harmonic Distortion Vs. Level
- Wow & Flutter; noise; weighted or flat
- Channel Separation 20Hz - 20k
- Delta Speed & Drift

Because of the modular plug-in design, the 1500A is designed to grow with you. The first plug-in option will be available soon: a spectral noise and flutter card. Other current accessories include a hard-copy printer, flight case, rack mounting ears, and our own test record for phono cartridge analysis.



Who Can Use It?

Broadcaster. Recording studio. Film sound studio. Audio manufacturer. Audio dealer. Service technician. Researcher. Virtually anyone whose job requires accurate evaluation of audio equipment performance. Wherever you are in the audio spectrum, it can make life a whole lot easier.

Clean up your act with the 1500A. It's intelligent. And so is a phone call to Sound Technology. We'll be pleased to send full information on the 1500A and our other industry standard test equipment.

Please send me more information on how the Sound Technology 1500A System can help me clean up my act.

NAME _____

COMPANY _____

ADDRESS _____

CITY _____ STATE _____ ZIP _____

PHONE _____

**S SOUND
T TECHNOLOGY**

Department 2003102, 1400 Dell Avenue, Campbell, CA 95008
(408) 378-6540 TELEX: 357445

TEST SET THAT YOUR ACT.

db New Products & Services

CASSETTE COPIER



• The Telex Copyette 1&3 is an economical high-speed cassette copier capable of copying three one hour cassettes in less than two minutes in a single pass. No special training is required to operate the Copyette 1&3. The operator simply inserts the original cassette and from one to three blank cassettes, then presses a single button. All cassettes automatically rewind to assure that all cassettes are copied from the beginning. Copies are then made of both audio tracks of the original on a single pass. When copying is completed, all cassettes again automatically rewind to the beginning.

*Mfr: Telex Communications, Inc.
Circle 50 on Reader Service Card*

WANTED

A magnetic recorder with a frequency response of at least 100 Hz to 10 KHz (-4db), with a signal-to-noise ratio of 20 db at 1 KHz. DC record bias and erase. Head life must exceed 5 hours per pole piece. Wow and flutter should be lower than 10%, with a tape speed of not more than 3 feet per second and a record time of at least 30 minutes.

Sound ludicrous? If db had published 50 years ago, the beginning of the steel tape era, such a want-ad would have also appeared farfetched, because of its outrageously high specs!

We've come a long way since the "Blattnerphone" steel band recorder of 1932: the 1936 German Magnetophon audio tape recorder (the world's first modern hi-fi magnetic recorder), the American wire recorders of Marvin Camras' Armour Research Institute, Harold Lindsay's 1947 Ampex Model 200A audio tape recorder, 3M's outstanding developments in both audio and video tape since 1946, to name just a few of our industry pioneers.

The Museum and Archives of Magnetic Recording, an industry-supported project to preserve our recording heritage, wants to hear from you about the whereabouts of old wire and tape recorders (audio and video), old tape recordings, photos, and documents.

Please contact

PETER HAMMAR, Curator

**MUSEUM AND ARCHIVES OF
MAGNETIC RECORDING**
c/o Ampex Corporation
401 Broadway, Redwood City, CA 94063
telephone (415) 367-3127

GRAPHIC EQUALIZER

• The GE-3 graphic equalizer, like its predecessor the GE-2, has a combination of octave and one-half octave filter bands. The GE-3 provides the equalization capabilities of a one-half octave filter set at frequencies below 250 Hz. Above 250 Hz, bands are set on octave centers. This combination of octave and one-half octave equalization provides high resolution in the low frequencies where EQ problems are likely to occur. On the rear panel are the input and output connectors that allow the GE-3 to interface with a variety of sound systems. The GE-3 provides both balanced and unbalanced inputs as well as outputs. The balanced connections are made via standard XLR type jacks, while the unbalanced are made via 1/4-inch phone jacks. Both the unbalanced and balanced outputs are capable of driving load impedances of 600 ohms or greater to full rated output levels without distortion, and all inputs and outputs are protected against potentially damaging levels or loads.

*Mfr: Cerwin-Vega
Price: \$550.00
Circle 51 on Reader Service Card*



COMPRESSION DRIVERS

• Designated as EAS-45HD05EX and EAS-60HD05EX, these new drivers are designed to improve reliability and acoustical performance in professional speaker systems. The compression drivers will withstand high voice-coil heating while exhibiting an improvement in output power linearity. Among the features of the drivers: high power handling capability (with ferrofluid); titanium diaphragm for long life; rhombic edge; multi-slot phasing slug, and a high efficiency magnet structure.

*Panasonic Company
Circle 52 on Reader Service Card*



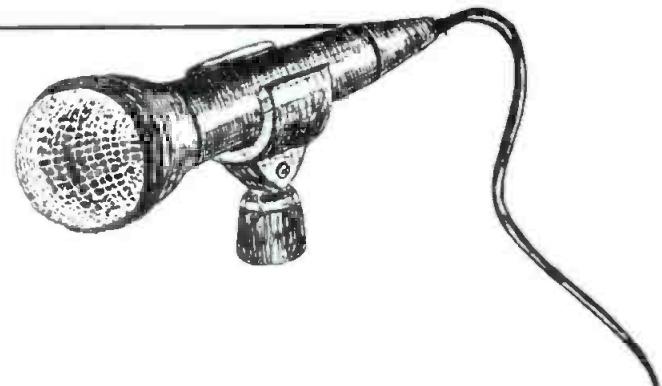
MULTITRACK TAPE RECORDERS

• The SCM 381 multitrack tape recorders are available in 8 track and 16 track one-inch formats, and in 16 track and 24 track two-inch formats. These machines feature very low wow and flutter, low noise, and small size. Full remote control, varispeed and search to zero facilities are standard, as are separate sync outputs on each track.

*Mfr: Soundcraft Electronics
Circle 53 on Reader Service Card*



SPECIAL PRE-PUBLICATION OFFER



announcing:

The Microphone Handbook by John Eargle

If you work with microphones, you need this up-to-date book.

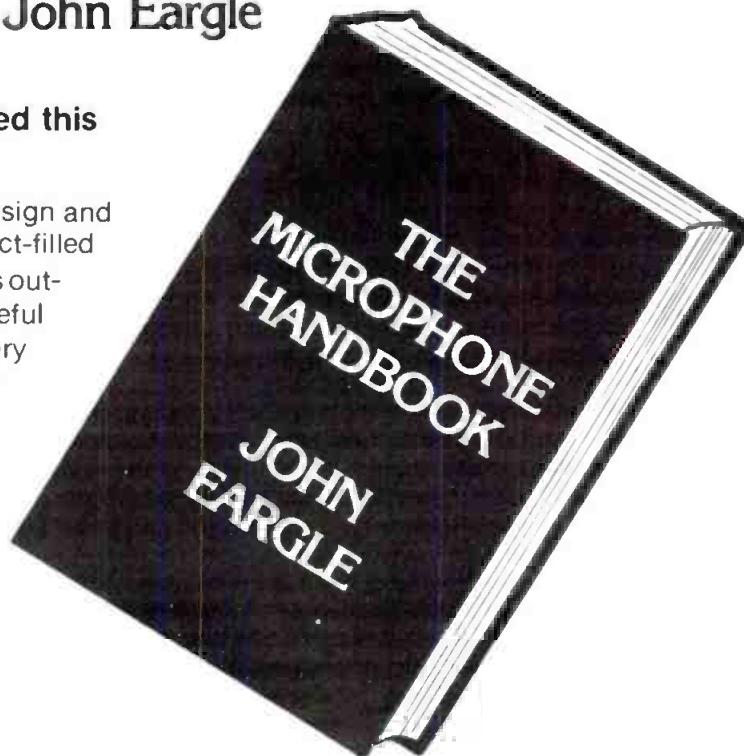
At last, the practical aspects of microphone design and usage are explained in a new and concise fact-filled volume, prepared by one of professional audio's outstanding experts. This book is so full of useful information, you will find yourself using it every time you face a new or unusual microphone problem.

Among the fact-filled chapters:

- directional characteristics—the basic patterns
- using patterns effectively
- microphone sensitivity ratings
- remote powering of capacitors
- proximity and distance effects
- multi-microphone interference problems
- stereo microphone techniques
- speech and music reinforcement
- studio microphone techniques
- microphone accessories

and more.....

The Microphone Handbook by John Eargle will be published in late spring 1981, at a list price of \$28.50. However, you can reserve a copy at the special pre-publication price of \$22.50, including postage. During this special offer, no Visa or Master Charge orders can be accepted. The expected publication date is June 1, 1981.



ELAR PUBLISHING COMPANY, INC.
1120 Old Country Rd., Plainview, NY 11803

Name _____

Address _____

City/State/Zip _____

My check for _____ copies of THE MICROPHONE HANDBOOK at \$22.50 each is included. NY State residents add appropriate tax. Full refund within 15 days if dissatisfied and book is returned in good order.

THIS OFFER EXPIRES JUNE 1, 1981

Outside U.S.A. add \$2.00 for postage.

Checks must be in U.S. funds drawn on a U.S. bank.

EXT MONTH, the topic at the fourteenth annual Midwest Acoustics Conference will be "Electronic Modification of Musical Sounds." According to a recent press release, conference topics will include: characteristics of musical sounds, history of electronic sound modification, electronic sound modification for the performing musician, time domain and frequency domain sound modification, nonlinear modification of sound, synthesizers, and a look at future trends. For time-and-place details, see the MAC ad on page 10.

And for still more on the subject, see this issue of db, where we begin our two-part series on "Music in the Studio: Electronics & Acoustic."

Electronic music has been around almost as long as electronics, though many of us in the recording studio think of it as something invented by Dr. Robert Moog and Walter (now Wendy) Carlos. The Carlos "Switched-on Bach" album brought electronic music out of the laboratory and into the studio, where at first it created a sensation. But it soon faded into a semi-retirement of sorts, as listeners grew bored with the "silly piano" antics of the legion of Carlos imitators.

Within recent years, electronic music has come back again, and this time it is apparently here to stay. The novelty aspect seems to have worn off, and the electronic music synthesizer is now taken seriously as a creative tool on stage and in the studio.

The synthesizer is now a likely candidate for a piece of next year's studio equipment budget. Its versatility makes it an attractive client pleaser—especially if there's someone around who understands how to make the damn thing work!

The vocoder is perhaps the prime example of a synthesizer that needs to be understood. Given the right inputs, it can do wonders; with the wrong inputs—nothing! Since one input will be used to modify the sounds of another input, it follows that the two input signals must bear strong spectral and temporal relationships, but that's another story. Specifically, its Felix Visser's story. This month, author Visser gives us a detailed look at the vocoder's insides, and helps unravel some of its mystery.

Electronic music brings with it the capability for extending the limits of the frequency spectrum of reproduced music. According to Michael Rettinger, electronic music and digital recording technology require improvements in studio monitoring systems, to meet the demands of a wider-range frequency response. At low frequencies, better response means moving more air; for high-energy output at 25 Hz, as many as eight 18-inch woofers may be required. Taking note of the physical impracticality of this, Rettinger proposes a ceiling-mounted folded horn system that cuts down on the number of speakers required, and enables us to construct such a system within a reasonably-sized control room.

More often than not, the synthesizer is tuned to the traditional well-tempered scale given to us by Bach (Johann Sebastian, not W. Carlos). But why not a metric scale to suit the needs of the metric movement? Why not, indeed? The synthesizer makes it easy enough, as Hal Lion and Jim Fox discovered during a recent recording project for the US Metric Board. Once they figured out what a metric scale should look like, it was easy enough to hear what it would sound like on the synthesizer. Although there's little chance that metric music will sweep the country, it's just one more example of how the synthesizer may be put to work in the studio.

One of the earliest predecessors of today's synthesizer was Thaddeus Cahill's Telharmonium, which consisted of some 200 tons of alternators, dynamos, transformers, gear and shafts, plus a two-rank keyboard. Suitably impressed, Ferruccio Busoni wrote, in 1907, that "... music is born free, and to win freedom is its destiny."

In the early '80s, it's difficult to appreciate the "freedom" that comes with a 200-ton instrument. Needless to say, today's synthesizers are a bit lighter, and certainly a lot more sophisticated. The latest generation is of course built around the microprocessor, which weighs a lot less than any collection of gears and shafts around. To find out just what we can expect from these grandchildren of the Telharmonium, Ralph Hodges visited a few west coast manufacturers, to bring us up to date on the latest synthesizer hardware (and software).

For still more information on electronic music, just check in with any of the manufacturers in our synthesizer directory also found in this issue of db.

Vocoders

The secrets of the mysterious vocoder revealed.

IF THERE WILL ever be a list of electronic audio instruments which have caused confusion in the professional audio and music industry, the vocoder will definitely be on it. Recently, the mysterious vocoder has become a desireable, though costly, recording studio instrument. However, the mystery element persists, often because the actual purpose and musical versatility of the vocoder are somewhat—or often, totally—misunderstood.

THE BASIC PRINCIPLE

Though it is hardly possible to list all efforts made in history to synthesize human speech, the name of Homer Dudley is inseparable from today's vocoder. In 1936, he patented his apparatus to analyze and remake speech, which he called a Vocoder, because it was based on the principle of coding the voice and then reconstructing the voice in accordance with that code. (So please, no more "Vocorder"—it has nothing to do with recording the voice!)

The principle of the Dudley vocoder, where speech analysis is performed by a set of parallel-band filters, is commonly called the channel vocoder, as opposed to the formant vocoder, where the resonances produced by the oral and nasal cavities are simulated by several tunable-formant or resonance filters. Being merely a speech synthesizer, instead of an analyzer/synthesizer combination, the formant vocoder is less interesting for music applications and therefore will not be discussed further in this article.

Basically, the channel vocoder can be divided into two main sections: the analyzer and the synthesizer, as shown in FIGURE 1. Each consists of identical filters covering a specific frequency range. The analyzer filters divide the speech spectrum into narrow bands, from each of which a voltage is derived. The voltage is proportional to the energy in the band.

In the synthesizer, a second group of filters divides the spectrum into the same narrow bands, and is followed by an amplitude-controlling device, such as a VCA (voltage-controlled amplifier or attenuator).

By connecting all analyzer control-voltage outputs to the respective VCA control-voltage inputs in the synthesizer section, the speech spectrum can be imposed upon a carrier

signal (a musical instrument, for instance) which is fed into the paralleled audio inputs of the synthesizer's filter bank. This creates the almost-classic "talking music" effect.

A similar effect can be achieved by the so-called "mouth tube" or "mouth bag." Here, the carrier sound is acoustically injected into the speaker's mouth by a flexible tube, and thus can be more-or-less articulated. Quite apart from unpalatable side effects, such as jaw-muscle spasms, this is a crude way of imposing speech upon another sound.

It should be made clear that the vocoder deals only with harmonic structure, and in no way is concerned with fundamentals, such as the pitch of the voice to be analyzed. Therefore, a circuit which can be very useful when using the vocoder as a voice synthesizer/processor is the pitch-to-voltage converter.

The only pitch-determining component is the fundamental of the carrier sound (the artificial vocal cords). When this pitch is changed, the pitch of the voice will change accordingly. When the pitch of the carrier is kept constant, a change in pitch of the real voice will only result in a different timbre of the synthesized voice. So, in order to generate complete, melodic speech, a device will be needed to change the pitch of the carrier in accordance with the pitch of the real voice. Such a device could be a pitch follower/extractor, as it is generally called. By changing the ratio between real-voice pitch and conversion factor, either linearly or non-linearly, many unrealistic but interesting artificial voice effects can be obtained.

THE ANALYZER

In the Syntovox 221 vocoder, which consists of 20 analyzer and 20 synthesizer channels, each analyzer channel can be split into three subsections: a band filter, a full-wave rectifier, and a low-pass filter with an LED readout. The combination rectifier/low-pass filter is also known as an amplitude demodulator or envelope follower. The analyzer section is shown in FIGURE 2A. The number 20 is arbitrary, and not a scientifically-determined minimum or maximum. (In the case of the Syntovox 221, it was partly inspired by the available standard matrix format chosen to interconnect the analyzer and synthesizer.)

If the vocoder is to be used exclusively to reconstruct intelligible speech, 15 to 16 filters are sufficient to cover the necessary frequency range from about 100 Hz to 3 kHz, with a resolution of 25 percent, which gives us a one-third octave filter spacing. This will provide a fairly-accurate picture of the speech spectrum. However, extending and subdividing the total frequency range beyond the 3 kHz limit will add more definition

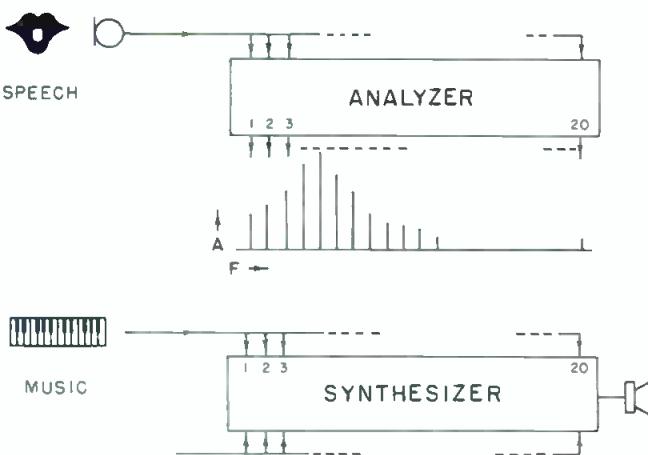


Figure 1. The basic building blocks of the vocoder.

and clarity, specifically where fricatives and sibilants are involved.

An ideal filter would exhibit a flat response within, and infinite attenuation outside, the pass band. In the Syntovox 221, a compromise has been found by giving the filters a relative bandwidth which is narrower than their one-third octave spacing, in order to achieve as little overlap as possible. Moreover, the filters—which are eighth-order—have a very rapid roll-off of 48 dB-per-octave, which increases to 54 dB-per-octave over the first octave.

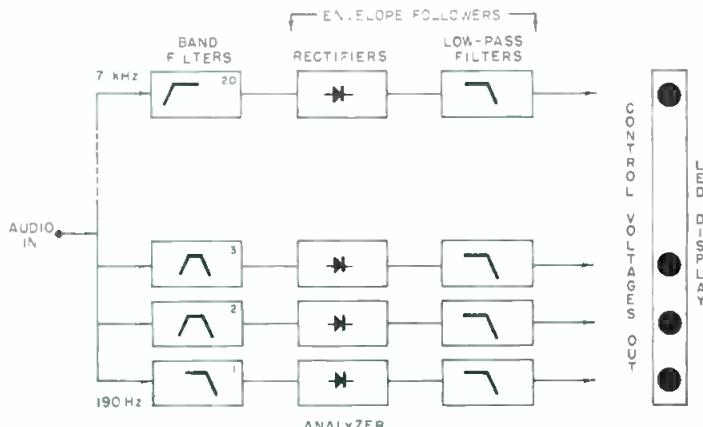
Although the dips in-between the narrow-band filters will affect the true response of the input signals to both analyzer and synthesizer, they greatly improve intelligibility and effect.

For designers of vocoders and speech synthesizers, it is unfortunate that the human ear is very sensitive to phase relations between spectral components. Frequencies, approximating the resonance frequency of a filter, will be subject to substantial phase shift, causing a different timbral perception, even though amplitude relations within the spectrum have been hardly affected. This phase shift side-effect will be more dramatic the higher the order of the filters, thus creating the unnatural speech effect that is a characteristic of the vocoder.

As with most man-made contraptions, the vocoder is full of compromises. But reducing the filter slopes, and thus reducing phase shift effects, would not result in a better vocoder. Large, overlapping areas create a blurred speech response, along with a poor effect response when the vocoder is used in music.

One more compromise in the analyzer section is the low-pass filter following the full-wave rectifier. In order to obtain rapid response of high frequencies and transients, this filter must be as fast as possible. On the other hand, a large ripple margin will definitely create intermodulation effects, which can be very disturbing.

Figure 2A. The Analyzer section.



The problem can be more-or-less resolved by adapting the cut-off frequency of the low-pass smoothing filters to the audio pass-band of the preceding analyzer filters. By choosing a roll-off point about ten times higher than the center frequency of the audio filter, an optimum can be found with respect to the phenomena of transient response and intermodulation as explained previously.

Finally, the LED control voltage read-out array was not only applied for its anticipated appeal—an expectation which has been met—but because it definitely has the function of displaying the spread, and to a certain extent the amplitudes, of the spectral components in the analyzed signal.

To cover the whole audio frequency range with a set of 20 filters, spaced at one-third octave intervals, would not be possible. Since the vocoder is only concerned with the harmonic structure of speech in the first place, it is not necessary to maintain this resolution either, so by designing the lowest and highest filters as low-pass and high-pass, the total audio range can be covered from about 20 Hz to 18 kHz.

THE SYNTHESIZER

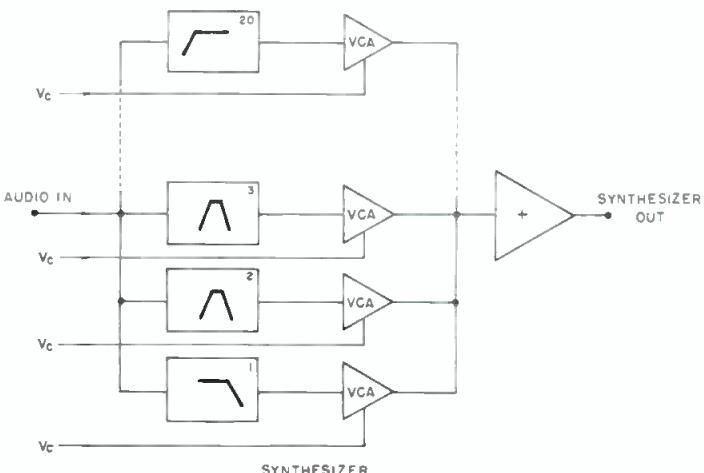
A description of the synthesizer filter bank will be easier, now that it is clear that the audio filter section is a replica of the analyzer filter bank. The only practical problem within a certain budget is the fact that both sections should be exactly the same, which means that only a small tolerance in component spread can be allowed. Any deviation from the analyzer center frequencies will cause unwanted formant shift.

In FIGURE 2B, the basic layout of the synthesizer section is shown. FIGURE 3 illustrates the performance of the filter bank in both the analyzer and synthesizer sections. The carrier sound is split up by the synthesizer filter bank and these signals, after being processed by the VCAs, are summed, and become the vocoder output signal. All control voltage inputs of the VCA bank are supplied with an attenuator followed by an LED readout. These attenuators can be used to "equalize" the vocoder effect.

The trickiest part of the synthesizer section is the voltage-controlled gain device following the filter. In order to copy the spectral image in accordance with the control voltage delivered by the analyzer, each synthesizer filter is followed (or preceded, as sometimes is done) by a voltage-controlled gain device allowing spectral components of the artificial vocal cords (the carrier) to pass through to the output of the vocoder. Such a gain cell could be a VCA, an OTA (operational transductance amplifier) or an electronic switch controlled by a pulse-width modulation (PWM) system.

The PWM's amplitude control is achieved by simulating a variable resistor in the audio path, by means of an electronic switch controlled by a high-frequency generator. This switch is alternately opened and closed, either by a narrow pulse whose repetition rate can be voltage controlled, or by a voltage-controlled duty cycle rate.

Figure 2B. The Synthesizer section.



The Ultimate PORTABLE RECEIVER for wireless microphones

now available with DYNEX option

The Cetec Vega Model 66 is the most advanced portable receiver in the industry. Designed for the professional sound user, the 66's compact size makes it suitable for mounting to leading portable recorders, both audio and VTR's. The unit, measuring just 5.4" W x 1.3" H x 6.25" D, operates from either internal 9V batteries or an existing DC voltage source.

For convenience, Cetec Vega now offers a portable system for multi-channel applications, the Model 151 Quad Case. This light weight, rugged case houses four complete Model 66 receivers, and has a built-in antenna splitter that feeds all four receivers from one detachable antenna. Power for the unit is provided by standard alkaline "D" cells or an external DC source.



Model 151 Quad Case

Model 77 Transmitter



Cetec Vega

Division of Cetec Corporation

P.O. Box 5348 • El Monte, CA 91731 • (213) 442-0782 • TWX: 910-587-3539

Circle 16 on Reader Service Card

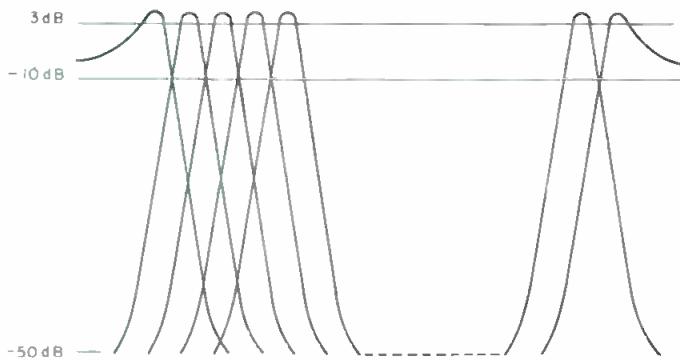


Figure 3. The response of the filters in the Analyzer and Synthesizer sections.

VOICED/UNVOICED DETECTOR

The analyzer/synthesizer sections provide a straight-forward vocoder system, with which fairly-intelligible speech and good musical results can be obtained. However, the addition of a voiced/unvoiced detector will add even more intelligibility and clarity to the synthesized vocal effects. FIGURE 4 shows a block diagram of such a detection system, to which many variations and additions can be made. The detector can discriminate voiced and unvoiced sounds by continuously comparing the energy in two different frequency bands. One band is 30 to 800 Hz, and the other is 2 kHz and up. The decision of the detector is based on the assumption that voiced sounds have less high-frequency energy than unvoiced sounds, which fortunately is true for almost all speech sounds. Almost, because composite sounds such as "V" and "Z" can create instability of the detector when pronounced over a long period of time. Under normal speech circumstances, the detector, responding very rapidly to changes of the energy in these two bands, will switch alternately between voiced and unvoiced, and neither the indecision nor the transitions will be audible.

In order to extend versatility in the electronic music studio, and with computer applications in mind, the detector should be equipped with control inputs and outputs, as well as an inhibit input when the vocoder is to be used with other triggering devices.

Another possibility to add intelligibility to the vocoder effect has been provided by Harold Bode, who patented the clever solution of adding the high-frequency end of the real speech spectrum to the vocoder output signal. FIGURE 5 shows the basic setup (U.S. patent 4,158,751).

To provide the necessary high-frequency spectrum for fricatives and sibilants without using an expensive detection system, noise may be added to those synthesizer filter channels that are above 2 or 3 kHz. The noise is constantly present, and it will feed through to the output when high-frequency components in the voiced spectrum open the high-frequency

Figure 5. The dashed-line path improves intelligibility by adding high-frequency audio components to the vocoder output.

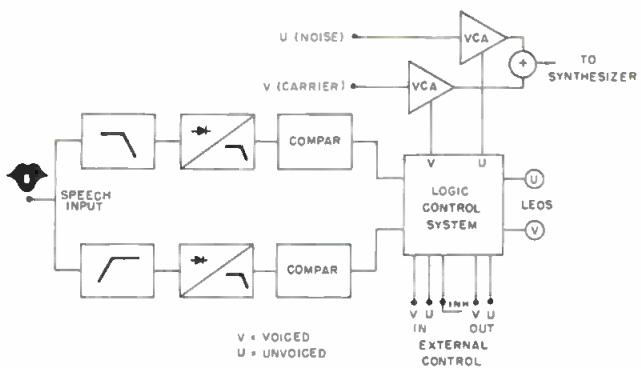
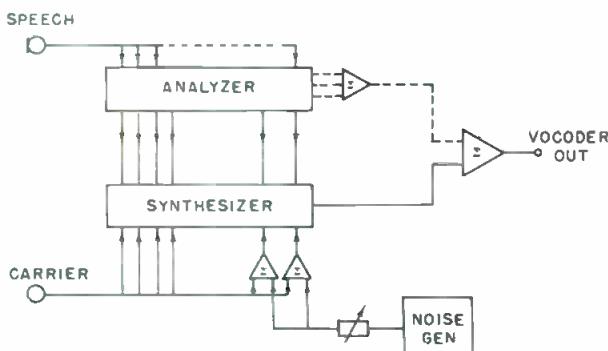


Figure 4. The Voiced/Unvoiced Detector.

VCAs in the synthesizer. In general, this solution offers better intelligibility of the synthesized voice effects.

CARRIER SOUNDS—THE ARTIFICIAL VOCAL CORDS

The carrier sound, which will replace the signal normally produced by the vocal cords, is subject to certain conditions which will be discussed later on. For speech synthesis, it is necessary to generate a carrier signal similar to that of the vocal cords, as shown by the functions in FIGURE 6.

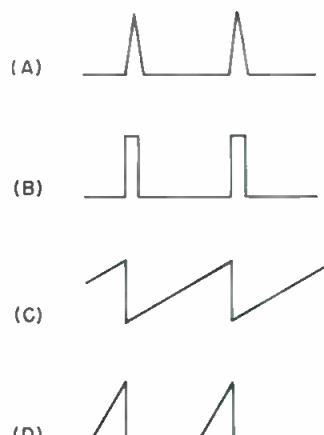
These waveshapes each give a different speech result because of their respective spectra. A very narrow pulse signal will provide a synthetic voice of piercing, rattling quality, because of its very strong high-frequency spectrum. A sawtooth and a spaced-sawtooth will give a more pleasant, mellow-sounding voice quality. The pitch generator should be controlled by external sources, such as pitch followers, envelope followers, low-frequency generators and random generators. All these control sources can take away much of the static, machine-like impression that vocoder speech usually makes. It is an interesting experiment to note that a voice will sound less artificial and even more intelligible when the pitch is modulated, instead of being kept constant.

When using external carrier sources, such as musical instruments, an automatic bypass circuit can be of great help to bypass or cross fade the carrier to the output when no analyzer input signal is available. Such a circuit has been incorporated in the Syntovox 221, and is called a "fill-in." Other manufacturers supply similar facilities, labelled "silence bridging" or "pause stuffing."

CONTROL VOLTAGE PATCHING

One aspect of the vocoder still to be explained is the interface between the analyzer and the synthesizer. As explained previously, and shown in FIGURE 1, all control voltages generated by the analyzer section have to be fed to the control voltage inputs of the gain cells in the synthesizer section.

Figure 6. Various carrier wave shapes which may be used to simulate vocal-cord sounds.



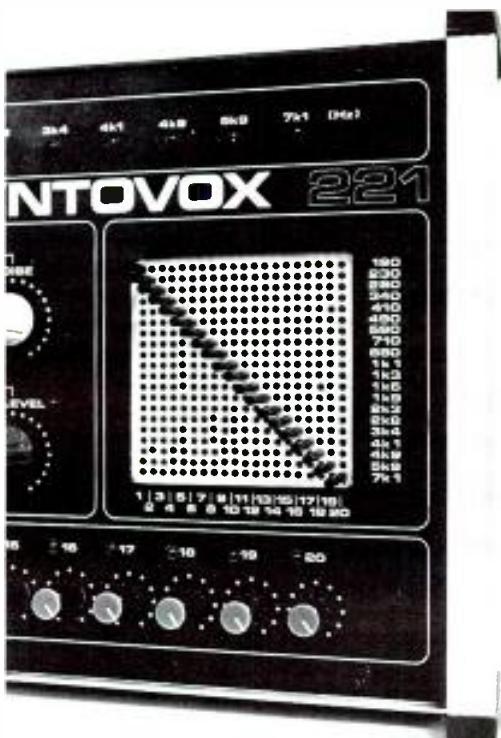


Figure 7. A 20-x-20 matrix connects the analyzer and synthesizer sections of the vocoder.

One of the advantages of not connecting them permanently is that both sections can be used to control, or be controlled by, external equipment. Also, this creates the possibility of connecting the analyzer outputs to other than their respective synthesizer channels. The problem of choosing a way to provide this facility is purely practical, and few manufacturers of larger vocoder systems have gone to this trouble. One way of providing a compact and versatile patching system is the matrix, which allows optimum freedom of routing control voltages. In addition, all outputs and inputs of the analyzer and synthesizer

should be available for use in electronic music studio setups.

FIGURE 7 shows the matrix which connects the Syntovox's analyzer and synthesizer sections. In FIGURE 8, all sections discussed so far are shown in a complete block diagram.

One of the most obvious applications of the matrix is formant shifting, which means that formants (typical resonance peaks) are transposed to frequency areas other than where they originate. This feature makes the vocoder an interesting instrument with which to generate different types of voices. Though it is often suggested that the sex of the voice can be changed by shifting formants and raising pitch, this is not true. However, the character of the voice can be changed dramatically, and a certain touch of different age can definitely be achieved.

APPLICATIONS

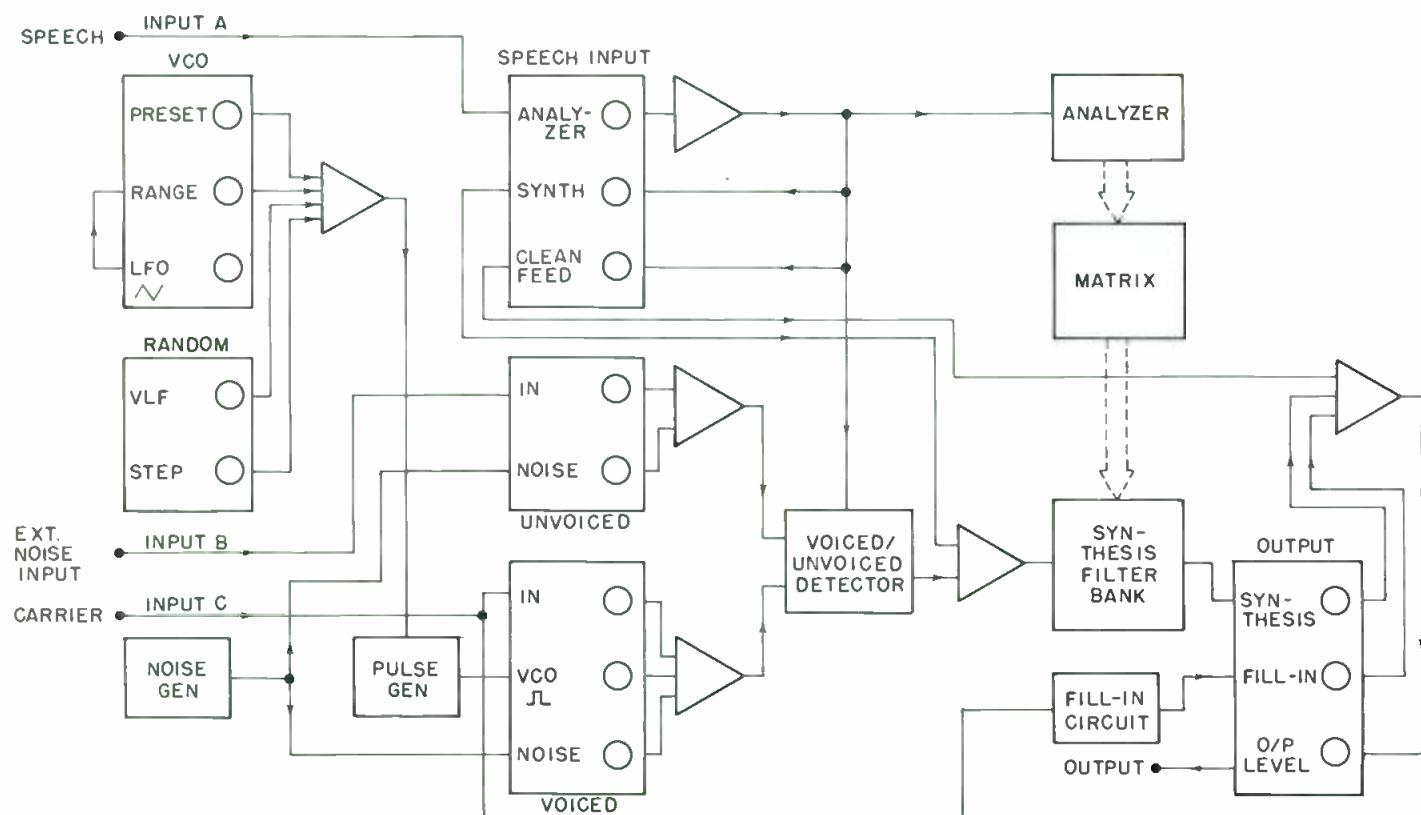
When marketing vocoders, one of the main problems encountered is that it is a typical two-input device. The significance of this is not always fully appreciated. In any demonstration, it is almost impossible to show the versatility of the vocoder, since every carrier creates its own effect, and every modulator creates a different effect with the same carrier.

The only thing which can really be made clear to interested people is some basic information on intelligibility, response, and general sound quality.

Japanese manufacturers have appreciated the two-input dilemma, and equipped most of their vocoders with some kind of keyboard instrument, producing organ-like or string-like sounds. They too must have realized that keyboard players belong to the happy few who can play and speak or sing at the same time. However, this conception can easily lead us to the conclusion that a vocoder is a typical keyboard instrument extension, which it isn't. Though it is true that in many cases the voice/keyboard combination gives the most recognizable response within a relatively short setup time, the vocoder makes it possible to use almost any sound source to modulate any other sound source.

There are restrictions, especially concerning frequency spectrum and synchronism between sound sources. In order to

Figure 8. A simplified block diagram of the Syntovox 221 Vocoder.



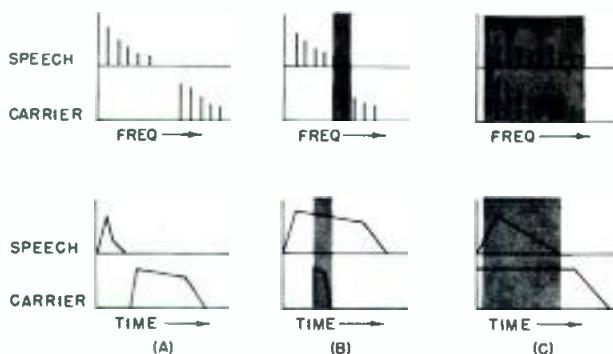


Figure 9. Optimum vocoder performance requires an overlap of spectral and temporal content. In other words, the carrier signal must be harmonically rich, and occur concurrently with the speech input signal. (A) The conditions shown will produce no effect. (B) The two narrow shaded areas indicate a slight effect, which will occur during a brief time interval. (C) Maximum effect is achieved, due to a complete spectral overlap, which occurs continuously.

get the best effect, it is necessary that both speech and carrier overlap spectrally and in time, which is graphically illustrated in FIGURE 9. A practical situation may make these examples more clear: a poor effect—or no effect at all—will be your reward for trying to modulate the sound of a bass drum with that of a flute, or vice versa. The vocoder user will also get into trouble when trying to impose speech upon a sine wave, or other pure sound.

Problems are bound to come up when modulator and carrier are of short duration and not in perfect sync. A recorded guitar track, producing a sequence of rhythmically limping chords, is not the right carrier to start with, unless the owner of the voice who is going to try to modulate the guitar sounds is alert enough to limp along in sync. Therefore, when a vocoder is used with previously recorded tracks the user may be disappointed with the results. The best way to create interesting vocoder effects is in a live situation, when there is artistic feedback between carrier and modulator. Truly, this is one of the most attractive sides of the vocoder. Being a real-time instrument, it can be used to instantaneously voice-control the timbral quality of sounds produced by electronic music instruments.

Here, the nature of the vocoder touches one of the marketing problems. Due to its complexity, the vocoder is often classified as a piece of high-technology hardware, to be sold to recording studios. On the other hand, the vocoder demands a thorough—almost musical—training of its user, which would make marketing aimed at the music industry seem the most logical way to go.

A sensible way to outline the possibilities of the vocoder may be by placing them in different fields of applications. In the first place, there is the scientific sector, where the vocoder can be a suitable instrument in speech or phonetic research. The effect of transposing formants can be simulated easily by connecting analyzer outputs to different synthesizer control inputs. Vowels can even be inverted by making partial cross patches on the matrix. A very interesting project at the Utrecht State University in Holland tested to what extent speech intonation is important to the intelligibility of the message. This was done by sampling real speech at a 10 kHz rate, and by changing the pitch of the synthesized voice under computer control, which allowed the intonation patterns to be varied every 2 milliseconds. Another purpose for which the vocoder can be used is speech training by producing properly-pronounced vowels and comparing the trainee's pronunciation of them. This can be achieved under computer control by digitizing the analyzer output information and comparing it to digitally-stored information.

A very appropriate application of the vocoder is in the field of animated film and sci-fi productions, making different alien voices by shifting formants and modulating the pitch generator in several ways. Since it is relatively simple to create numerous voice characters with the vocoder, it is strange to note what



Figure 10. The Syntovox 221 20-channel electronic effects Vocoder.

troubles sound engineers and producers go through, making their movie character voices with the help of ring modulators, filters, envelope followers and shapers, almost desperately trying to avoid the use of the vocoder.

It can be useful to process a guitar through spectrum-enriching effect devices such as boosters, fuzz boxes, doublers, etc. The same applies to other instruments generating more-or-less pure sounds. Once the carrier is rich in harmonics, it is possible to mold it into the timbral shape of vocalized sounds, sung or spoken into a microphone connected to the analyzer input. Now, dynamics and color are under complete vocal control and can be modified instantaneously due to the real-time nature of the vocoder.

On-stage use will require certain precautions, depending on acoustic conditions. Due to the phase shift introduced by the sharp filters, the vocoder can be very sensitive to acoustic feedback. Also, background noise can trigger horrifying effects. The classic trick with two reverse-phased microphones can be helpful, as well as directional, close miking. When close miking, attention should be paid with respect to inherent low-frequency boost, and when the microphone does not have an internal roll-off filter, conditioning with an external equalizer or by means of the attenuators at the synthesizer control inputs will be necessary.

Applications in the electronic music studio are almost unlimited, as long as the vocoder is an "open" system, meaning that the user will have access to everything controllable inside the unit. With the help of the analyzer section, frequency-dependent triggering or modulating effects can be obtained. A frequency-controlled band-pass filter is easily realized by sweeping a sine wave through the analyzer band. Interesting percussion effects can be generated by applying random sequences of sine bursts to the analyzer and by feeding pink noise into the synthesizer filter section. Depending on the positions of the patch pins in the matrix, intricate rhythmic structures by a multitude of percussive instruments can be obtained. In all, the vocoder can be used for a long list of effects which do not at all resemble the classic vocoder effect.

At least for the vocoder, the expression, "Your imagination is the limit," is not hype. ■

References

- "Analog Speech Encoder and Decoder"; Harald E. W. Bode; U.S. Patent No. 4,158,751; filed June 19, 1979.
- "Fundamentals of Speech Synthesis"; Homer Dudley. *Journal of the Audio Engineering Society*, Volume 3, No. 4, 1955.
- Phonetics (Fonetiek)*; Dr. L. Kaiser; 1964.
- The Use of the Elektor Vocoder*; Felix Visser; Elektor, September, 1980.
- Vocoders Today*; Felix Visser; Elektor, December 1979.

MICHAEL RETTINGER

Reproducing Electronic Music in the Control Room

Real advancement in the control room means increased low-frequency recording capability—not increased gadgetry.

THE SOUND RECORDING business is a highly-competitive one. When a studio can offer blue-sky novelties like Plasmatronics loudspeakers (\$6500 a pair), bass traps (\$2000 each, if space is available in the studio), helium-filled balloons for greater sound dispersion along the ceiling (only \$500 per balloon), the more-fickle clientele may become attracted to that recording facility.

But such devices are not really effective means of innovation or improvement; they are more in the nature of gadgetry and interesting exotica.

This writer believes that a *real* advance in sound recording and reproduction can be made by extending the reproducible low-frequency range of the program. There are at least two arguments in favor of this type of advancement: synthesizers can now generate bass notes lower than the more-traditional musical instruments, and, digital recorders can record and reproduce such notes with a much-greater signal-to-noise ratio than the usual analog recorder.

For good reproduction of low bass notes, the speaker's cone suspension must have a low resonance frequency. In addition, for large air-volume displacements of the loudspeaker diaphragm, the cone travel from its mid- or rest-position may often be more than 0.5 inches (1 inch total excursion). This tends to result in distortion which can readily be heard, because the ear's sensitivity increases with frequency in this range of the audio spectrum.

FIGURE 1 shows the loudness level contours below 100 Hz in a binaural free-field environment. For a loudness level of N phons, the sound pressure level (SPL) may be found by the formula

$$SPL = 160 - (67.5 - 0.4N)\log f.$$

As an example, at 25 Hz, when the loudness level is to be 100 phons, the sound pressure level is 121.5 dB, relative to 0.0002 microbars.

The cyclic air volume displacement of a circular piston in an infinite baffle may be calculated by the formula

$$V = \frac{0.0004(R)10^{SPL/20}}{f^2}$$

where

V = air volume displacement, in cubic feet-per-cycle,

R = listening distance, in feet,

SPL = sound pressure level, in dB, and

f = frequency, in Hertz (cycles-per-second).

With the frequency and sound pressure level given above, and a listening distance of 13 feet, the formula gives us a displacement of 1 cubic foot (1728 cubic inches)-per-cycle. At a frequency of 25 Hz, this means an air-volume displacement of 25 cubic feet-per-second.

Mr. Rettinger is an acoustics consultant and author of "Acoustic Design and Noise Control."

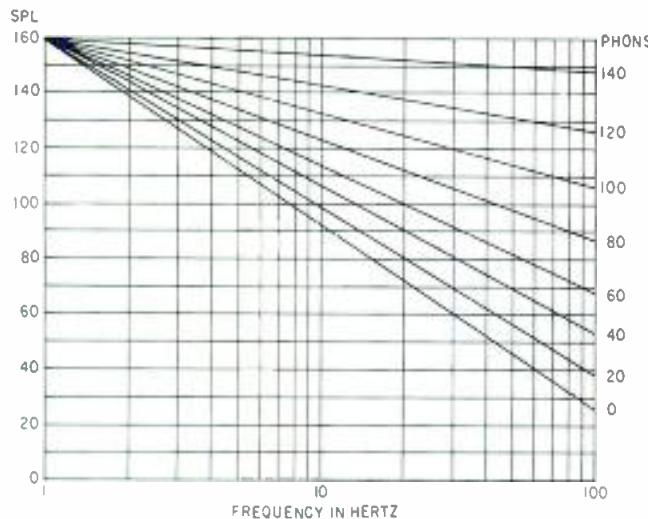


Figure 1. Idealized contours of equal loudness below 100 hertz.

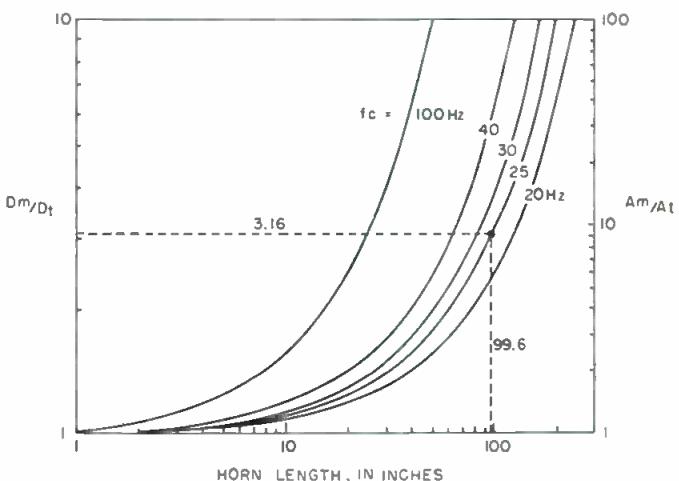


Figure 2. Horn length, in inches.

A loudspeaker with an 18-inch diameter cone has an active diameter of 16 inches, and therefore an active circular area of 201 square inches. Given a cone travel of 1 inch, we will require 8.6 speakers ($1728/201$) to produce the required displacement.

To avoid distortion in the reproduction of such low notes when the baffled diaphragm is driven to full excursion, an exponential horn should be used. Such an acoustic transformer can increase the efficiency of the reproduction system several times, thus requiring much smaller cone displacements (that is, fewer speakers) at full output. There is indeed a close relationship between speaker efficiency and distortion, particularly when impedance matches are involved.

An exponential horn acts as an impedance-matching device, acoustically coupling a heavy cone to the relatively-light air medium. It is not unlike a step-down transformer, coupling a high-impedance source to a low-impedance load. When the ratio of the horn's mouth-to-throat diameters is 3.16:1, the sound pressure at the mouth—like the high voltage at the amplifier output—is decreased by 3.16. However, the volume of air flow at the mouth—like the current through the loudspeaker coil—is increased by 3.16. When the horn is well made, there is no power loss along its length by acoustic energy absorption in the horn walls. The acoustic power along the horn length remains constant, because it is given by the product of sound pressure and air particle velocity, which remains constant. Note that a ratio of 3.16 for horn mouth-to-horn throat diameter corresponds to a ratio of 10 for the relative areas of mouth and throat, as long as the cross-section of the horn is either round or square.

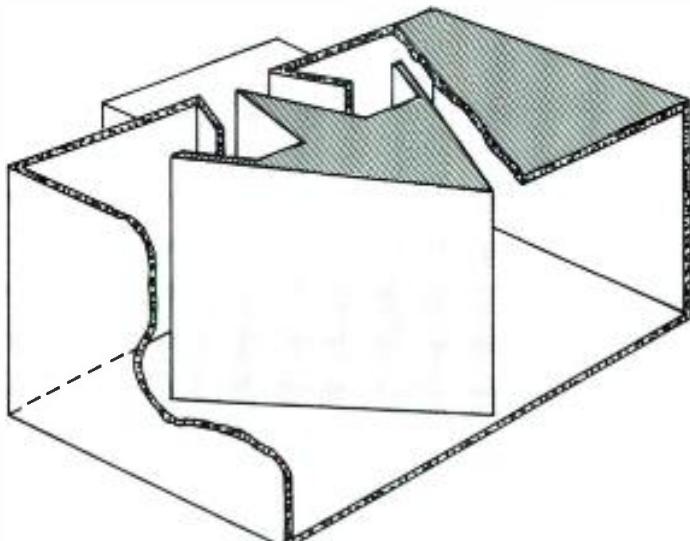
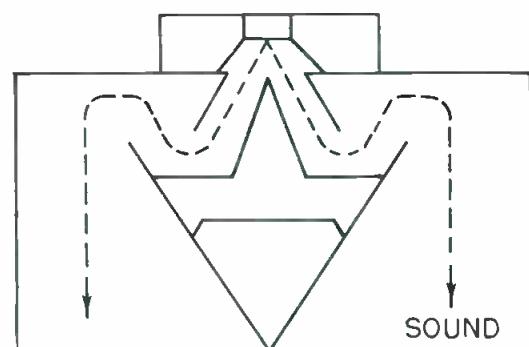
How long must an exponential horn be, which is to have a cut-off frequency of 25 Hz and a diameter ratio of 3.16? The equation is

$$\begin{aligned}
 L &= \frac{\log D_m - \log D_t}{0.217m} \\
 &= \frac{414.9 \log(D_m - D_t)}{f_c} \\
 &= \frac{414.9 \log(3.16)}{25} \\
 &= 8.29 \text{ feet}
 \end{aligned}$$

(This formula is explained in detail in the author's "Restricting Sound Radiation" in the October, 1980 issue of *db*—Ed.)

FIGURE 2 is a graph of horn length (L)-versus-diameter ratio (D_m/D_t), for various cut-off frequencies. The right-hand vertical axis also gives the mouth-to-throat area ratio A_m/A_t .

Figure 3. A folded low-frequency horn.



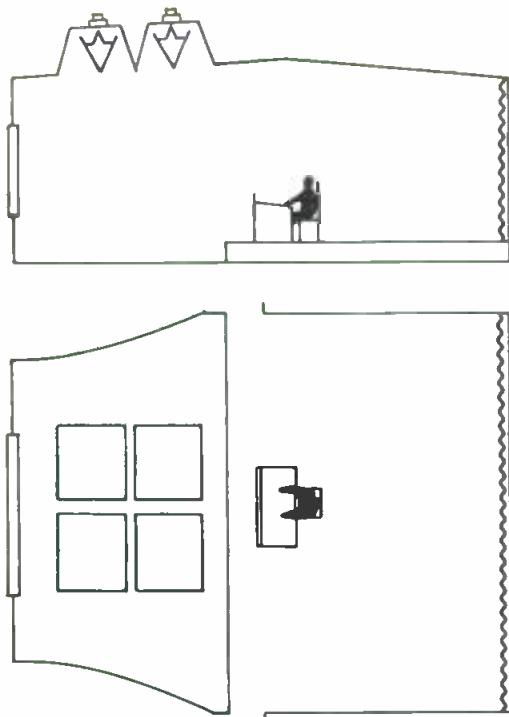


Figure 4. Low-frequency horns installed in Control Room ceiling.

A horn with a length of about 8.3 feet (99.6 inches) would probably have a total length of 10 feet, once the woofer is installed at the throat. To shorten this dimension, a folded horn may be used. FIGURE 3 shows such a unit, as was employed for the reproduction of those very low notes in Universal Studios' "Sensurround" system. The system is intended to reproduce not only these low-frequency sound waves, but also wind rumbles and other below-25 Hz effects.

FIGURE 4 shows the floor plan and elevation of a recording studio control room designed to accommodate the horn system just described. To facilitate construction, the horn mouths may be either square or rectangular. Because of the frequencies that are reproduced, these units are practically omni-directional, and need not be installed in the front wall of the room.

If digital recording technology eventually allows us to routinely record very-low frequencies, perhaps the audiophile listening room of the future will also see the installation of such horn systems, when (and if) these sounds make their way to the lp record. ■

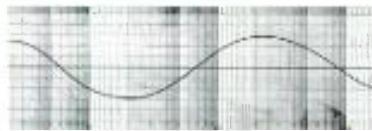
References

The best sources of information about loudspeakers and sound reproduction are the books by H. F. Olson. His first book "Applied Acoustics" has become a rarity because it is out of print. It was originally published by P. Blakiston's Son & Co. Inc., Philadelphia. It was written in collaboration with Frank Massa, a Swope Fellow at MIT. Two other books by H. F. Olson are "Acoustical Engineering" and "Modern Sound Reproduction," both published by Van Nostrand Co. Inc. in Princeton, New Jersey.

PRO-EQUALIZER by Soundcraftsmen

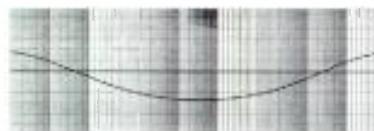


SECTION A



ACTUAL SWEPT-FREQUENCY RESPONSE CHARTS SHOWING THE EXACT CURVES RESULTING WHEN CONTROLS ARE SET AS SHOWN IN PHOTO ABOVE.

SECTION B



DESCRIPTION Exclusive Zero-Gain L.E.D.'s provide instantaneous in-out balancing convenience and optimum EQ performance. They are operated through a high precision Differential Comparator circuit with a read-out accuracy of 0.1dB, to provide true equalization with no change in the signal level being processed.

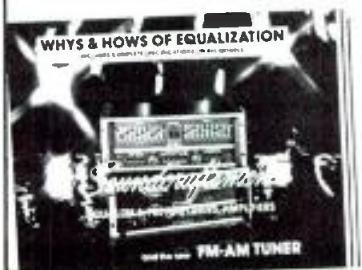
A filter Q of 2.0, combined with 3dB per octave slope and minimal phase shift, enables smooth reproduction of music without the sometimes harsh characteristic associated with a higher Q. A high Q can cause sharp dips between filters, sharp peaks at the filter centers, and pronounced phase shift.

Soundcraftsmen's expertise in the high fidelity field is combined with necessary shelving to improve speech and intelligibility, external

loop, pre/post EQ capability for external loop and balanced input/output capability. This combination provides all of the advantages of high fidelity specifications plus the practical features necessary in professional, commercial, public address, and industrial fields of operation.

FEATURES: FREQUENCY ANALYZER TEST RECORD, with Reference Tones, Test Tones, all-band Pink Noise, and complete instructions, is included . . . COMPUTONE CHARTS, for INSTANT-MEMORY Programming on full-size charts, for cut-out or marking of desired curves, are included . . . L.E.D. INDICATION of unity gain within 0.1dB . . . LINE INPUTS/OUTPUTS duplicated on front . . . PRE-EQ (LOOP) inserts EQ either pre or post external Loop/Monitor . . . 600 OHM BALANCED or unbalanced operation.

Six Graphic EQ Models. 1/3 and 1 octave. **\$249 to \$550**



FREE! 16-page Full-Color Brochure

Includes TEST REPORTS, complete specifications, Class "H" amplifier ENGINEERING REPORT, EQ COMPARISON CHART, and the "WHYS & HOW'S" of equalization—an easy-to-understand explanation of the relationship of acoustics to your environment. Also contains many unique IDEAS on "How the Soundcraftsmen Equalizer can measurably enhance your listening pleasures," "How typical room problems can be eliminated by Equalization," and a 10-POINT "DO-IT-YOURSELF" EQ evaluation checklist so you can FIND OUT FOR YOURSELF WHAT EQ CAN DO FOR YOU—SEND \$6.00 FOR EQUALIZER-EVALUATION KIT: 1-12" LP TEST RECORD, 1 SET OF COMPUTONE CHARTS, 1 COMPARISON CONNECTOR, 1 INSTRUCTION FOLDER

SOUNDCRAFTSMEN INC., 2200 So Ritchey, Santa Ana, CA 92705 • (714) 556-6193 • CANADA: E S Gould Marketing, Montreal H4T 1E5

Circle 35 on Reader Service Card

HAL LION and JIM FOX

Metric Music

In a series of radio news features introducing America to the metric system, what could be more appropriate than having "metric music" playing in the background?

HOW DO YOU MEASURE TAPE TENSION?

If something SOUNDS FISHY it may be your fish scale approach to measuring tension



The Tentel Tape Tension Gage is designed to diagnose problems in your magnetic tape equipment. Virtually all recorder manufacturers use and recommend the TENTELOMETER® for use with their equipment.

The TENTELOMETER® measures tape tension while your transport is in operation, so you can "see" how your transport is handling your tape, high tension causing premature head and tape wear, low tension causing loss of high frequencies, or oscillations causing wow and flutter. Send for the Tentel "Tape Tips Guide". The T2-H20-ML sells for \$245 - complete.



1506 DELL AVENUE
CAMPBELL, CALIF. 95008
(408) 379-1881
1-800-538-6894

Circle 21 on Reader Service Card

THE U.S. METRIC BOARD has the unenviable task of introducing the general public to the "delights" of the metric system. So, when the agency told us they wanted background music for a series of radio news features, we asked ourselves, "What does a ten-tone musical scale sound like?" We decided to find out.

Since the metric system divides everything into tens, we reasoned, why not go all the way with this thing and make the music metric too? After all, electronic technology being what it is, there's no reason to stay tied to the traditional twelve-tone scale; the question was, where to get some guidance for this excursion into the unknown? We found the frequencies of the conventional scale in *The Audio Cyclopedia*, but did not see any explanation as to where these numbers came from. So, we got out the log tables and the calculators, and with the help of a little algebra devised our own formula. It worked very well, and we later found it verified in a textbook chapter on musical scales.

If you've forgotten—or never learned—the mysteries of musical scale construction, here's how it's done.

The musical scale is a geometric progression. For example, the lowest A on the piano is 27.5 Hz, and the A above it is 55 Hz. The one above that is 110 Hz and so on through 220, 440, 880, 1760, and 3520 Hz. In other words, to find the frequencies of any series of octaves, just keep multiplying by 2. It follows that to find the frequencies of the notes on the scale within the octaves, you need to multiply by a figure that will, for each frequency in the scale, give the frequency of the next succeeding note. For the conventional 12-tone, equal-tempered scale, that figure is $\sqrt[12]{2} = 10^{\log 2/12} = 1.0594631$. When any number, f, is multiplied twelve times by 1.0594631, the final product will be 2f.

Hal Lion and Jim Fox are partners in Lion and Fox Recording, Inc., Washington, D.C.

To illustrate, using the octave between A-440 and A-880, begin by multiplying 440 by 1.0594631. This will give you the frequency of A#. Then multiply A# by the same number to get the frequency for B, and so on. The result is as follows:

A	440.000	E	659.255
A#	466.164	F	698.456
B	493.884	F#	739.989
C	523.251	G	783.991
C#	554.365	G#	830.609
D	587.330	A	880.000
D#	622.254		

Using the same principle for our ten-tone scale, we multiplied each frequency by $\sqrt[10]{2} = 10^{\log 2/10} = 1.0717735$. Starting with 440 Hz, this metric musical scale would be:

0	440.000	6	666.915
1	471.580	7	714.782
2	505.427	8	766.084
3	541.704	9	821.069
4	580.583	0	880.000
5	622.254		

Notice that Note 5, at the mid-point of the metric scale, has the same frequency as D#, at the mid-point of the conventional scale.

But rather than start with 440 Hz, we wanted an authentic, metric basis for our scale. We assumed a day divided into 100,000 "metric seconds" instead of the 86,400 conventional seconds and established a base frequency at 10 cycles per "metric second" (cpms). Then, of course, we had to make a concession to the fact that all of our equipment is calibrated in standard seconds. A quick calculation will show that 1 second = 1.1574074 "metric seconds," so that 10 cpms = 11.574074 Hz.

Beginning with this sub-sonic frequency, we kept multiplying by 2, until we reached the two octaves between 185.185 Hz, 370.370 Hz and 740.740 Hz (corresponding to 160, 320, and 640 cpms) that were within a tonal range suitable for our purpose. The metric octaves ("octaves"?—Ed.) up through the range we used are as follows:

cpms	Hz
10	11.574074
20	23.148148
40	46.296296
80	92.592592
160	185.185185
320	370.370370
640	740.740740

These tones, incidentally, correspond almost exactly to F# on the conventional scale. The middle tone in the octaves we chose to work with (370.370 Hz) corresponds to the F# above middle-C (369.994 Hz). Having identified the octaves, we then used the procedure outlined above to make the 10-tone scale.

The next step was to find a way of producing these sound intervals on available equipment. For this task, we turned to our resourceful synthesizer player, John Abernathy. He set the VCF (voltage-controlled filter) of his Arp Odyssey synthesizer to self-oscillate. He then patched the keyboard into the VCF and, taking a range of twenty notes, adjusted the VCF attenuator to "squeeze" them, so to speak, into an octave. When this was done, middle-C and the G# above C-above-middle-C sounded just one octave apart. This adjustment actually produces a 20-tone scale, from which the 10-tone scale can be realized by playing every other note: C, D, E, F#, G#, A#, C, D, E, F#, G#. If metric music ever catches on, one can of course devise an appropriate keyboard, perhaps with some color coding to distinguish octaves and half-octaves (Notes 0, 5, 0, 5, etc.).

When the scale had been established by adjusting the attenuator, the VCF cut-off was adjusted to bring it into line with the desired base frequency. A sine-wave audio generator was used, first to establish the base frequency, and then to check the

other tones in the scale, verifying the intervals as calculated, and voilà, "metric music"!

Now comes the hard part—composing an acceptable melody on this admittedly weird scale, which of course has no harmonics to speak of and needs to be heard in a sequence of single tones. John Abernathy tried a number of sequences. For the most part, attempts to adapt familiar melodies from the conventional scale did not yield very interesting results, but he composed some melodies that used the strange intervals to advantage. We finally settled on four melodic segments of five notes each—since the number 5 seemed appropriately metric. These segments are noted below on the special ten-tone staff we have designed.



We put these segments together in various combinations, producing a melody that was then surrounded by some other interesting musical effects. The Metric Board accepted the final result and uses it in the background of "Metric Magazine," a series of radio news features that has been distributed to stations throughout the country. ■

PRECISION

MAGNETIC

TEST TAPES

STANDARD TAPE LABORATORY, INC.

26120 EDEN LANDING ROAD #5 HAYWARD CALIFORNIA 94545 • 415 786 3546

RALPH HODGES

Zoop, Beep, Brap, Broop

Welcome to the brave new world of digital synthesizers.

IT MAY SEEM like yesterday when *Switched-On Bach* was climbing the charts, but it's actually ancient history. Bob Moog is reported to be breeding horses in New England, Tomita is presumably dancing with snowflakes somewhere in Hokkaido, and the days in which you punched up a sawtooth wave from a glorified audio generator, added a little square for seasoning, and then ran the whole thing through a ring modulator are dead and gone. The generation of electronic music is overwhelmingly digital by this time, and even if you believe that this development is hazardous to your health and vitality, you cannot deny the logic and potency of the approach.

Actually, there are probably not too many things worth doing that an analog synthesizer could not somehow be designed to accomplish; and with luck and care, it might even be possible to keep noise build-up from swamping the composite. But the fact is that things are easier with digital in so many ways. There is hardware flexibility; a typical analog circuit may be limited to one function, whereas a digital circuit can often be persuaded to perform several. There is control interface; the sound-generating portion of a digital synthesizer tends to be computer-like, just as the control section is computer-like. They live in the same sort of time frame, have similar signal-handling characteristics and drive requirements, and they present the overall designer with only one discipline to master fully, instead of two. And of course there are the matters of cost, complexity, compactness, and portability, which only increase the probability that anyone involved in audio work is going to have a run-in with one of these contraptions sooner or later. What is this brave new world going to be like?

Ralph Hodges is a freelance writer/editor living in the San Francisco bay area.

THE NATURE OF THE BEAST

First of all, what is an electronic-music synthesizer, really? (For this discussion, we'll restrict the category to *digital* synthesizers, they being in the majority and very much on the rise.) Is it a musical instrument, a signal processor, a sound-effects library, an automated music-composition machine, a recording device, a multi-media effects coordinator, a glorified mix-down engineer and editor, or a mindless brat of an electronic nuisance? Today, it's likely to be at least a little bit of all these. Tomorrow, it will probably take on the scheduling and billing for sessions and arrange for the contractor to come in and clean out the fish tank. The device need have no real identity whatsoever. The specific functions it can routinely perform have, as a rule, been deliberately circumscribed by the designer, who is coming to realize that normal people have to have some limitations to fetch up against, lest they get lost in potentialities and intricacies.

As a beginning, let's look at a comparatively simple but not especially typical device: the LM-1, offered by Linn Electronics for a low-key (in this league) \$5,000. The LM-1 specializes in something that may come as a bit of a surprise: real music, and specifically real drum and percussion music. Virtually every sound it is capable of making originated with a real instrument and a microphone, the pick-up of which was digitized and stored in hard read-only memory as single beats of a kick, a snare, two toms, two congas, tambourine, cabassa, high-hat, cowbell, and a little bit of *et cetera*.

The machine puts these single recorded beats together in the repetitive pattern of your choice, working basically in—but not limited to—two-measure chunks of 4/4 time. Select any one of a hundred random-access memories on the numeric keyboard, hit record and play, then whack out whatever you please on the pushbuttons assigned to the individual instruments, in time to

the metronome beat the device will give you (at whatever speed you want, of course). Want more instruments than your two fingers can play at once? Store what you've got, repeat the original recording sequence, and, in effect, overdub. You need another time signature? Count it out as if you were playing the actual instrument, and the memory will give it back to you on demand. Your problem is that you can't play at all? Well, if you can manage even a crude approximation of what you're after, the L.M-1's time base will correct your timing errors, to the nearest 1/16th note as a matter of course, but to 1/32nd-note triplets if you elect.

After you've finished loading up two measures with as much racket as you can tolerate, what then? A chain function enables you to link up memories for a continuous rendering of perhaps ten minutes of fairly complex, ever-changing rhythm patterns before storage capacity is exhausted. You just keep punching the new patterns in, or drawing on those that have already been stored. Should you need still more storage, the L.M-1 is based upon control signals that even a lowly cassette deck can record and play, so these can go right out a jack on the rear panel and get stored on tape.

Needless to say, the L.M-1 will give you back anything you've stored on demand, at the press of a few buttons. It also lets you hop quickly around from memory to memory, changing or erasing anything you wish. But in common with the more complex devices we're getting to, it is not intended to duplicate the functions of a multi-track console. If you require some reverb, you must get the signal out and into an echo send and return. There are separate outputs for all the instrumental sounds provided, but they are intended to be panned, balanced, and equalized on the board (which incidentally might take up more inputs than your usual scheme of miking a real drum set). The outputs, like the signals themselves, are mono, if that makes a difference to you. On the other hand, the various percussion sounds can be continuously varied in timing and pitch independently—something that only technology like this can manage.

SCIENCE-FICTION LAND

Useful as the Linn device can be, it does not really qualify as a synthesizer, being limited and ultimately inflexible in the variety of burbles and bleats it can emit. The big jobs, coming in at approximately \$14,000 and going up rapidly from there (especially with options), can generate any sound you've ever heard or can imagine, render it up as rock, boogie woogie, or a Viennese waltz, play it forwards or backwards, alter it even further beyond comprehension as they do so, and even print it out on a sheet of music paper. They come with organ-type keyboards, standard or optional, and therefore are conveniently playable in real time. What can be achieved at the keyboard will give some idea of the flexibility available. Key velocity—the speed or force with which the key is struck—can be programmed to regulate parameters that include dynamics, attack, vibrato, glissando, portamento, sustain, damping, or a combination of these. The mere act of pressing a single key can cause a note to rise slowly, blossoming with a peacock's tail of unexpected overtones as it comes, and then decline abruptly to sustain with a more conventional harmonic structure but a sudden addition of vibrato, and finally fade away into an eerie-seeming-silence. At a demonstration of the Con Brios ADS 200, a quick tap of a key brought forth a clear, mellifluous bell tone that decayed naturally; sustained pressure on the same key made it die abruptly and dissonantly, as if the clapper had remained pressed against the bell. With a new program punched in, the same key provided only the raspy squeak of a saxophone-reed attack with a quick tap, but the full bloom of the instrument when the key remained depressed. In other words, totally opposite decay modes for the same playing action. But I suppose it makes a certain amount of sense, once you get to thinking the way synthesizer people think.

The Con Brio synthesizer really does work hard at relating to the uninitiated player programmer. It is almost completely integrated into a single chassis, with a two-rank organ keyboard

below, a control panel to the upper right, and a CRT and floppy-disc drive on the upper left. (Additions, many of them optional, include foot switches and pedals, an alphanumeric keyboard terminal, a printer that raps out a score of what you've done in conventional notation, a sub-chassis for additional hardware, and continually floppy-disc software.) The controls are intentionally labeled for comprehension by an analog person, and many of them are continuously variable knobs. Pushbuttons are illuminated only when the time has come to use them, and the CRT display makes an admirable attempt to keep you aware of what has happened through each step of the programming.

The Con Brio 200 is based on digital oscillators—64 of them (expandable to 256), which is said to be the current record number. Digital oscillators put out discontinuous pulses that are ultimately smoothed into a continuous waveform by a low-pass filter, but otherwise they present no obvious difficulties *vis-à-vis* analog oscillators, and they confer a lot of nice advantages. But how do you go about manipulating them?

Many will select a combination of the 160 preprogrammed wave patterns from the two floppy discs supplied by the manufacturer, and just begin making music. But the courageously adventurous user will attack an earlier point of the input. Basic tones can be built up from sine, square, triangle, sawtooth, pulse, and noise functions of the oscillators, singly or in mixtures. These are pushbutton-selected. Then additional oscillators are brought in for control signals to modulate amplitude, relative phase, frequency, and to provide additive spectral touches, with the CRT display acting as an envelope guide along the way. The end result is called a "voice." It's a hypothetical musical instrument, with its characteristics stored on floppy disc. Like the preprogrammed voices, it is indexed by number and can be called up for playing on the keyboard at the touch of a few numeric buttons; and it will never go away unless you choose to erase it.

But now it is necessary to play the instrument, and for that purpose a properly instructed keyboard musician is certainly

Figure 1. The Con Brios ADS-2000 Digital Music Synthesizer.



the best alternative. Besides the voice constructed above, as many as three others can be accommodated on the two-rank keyboard at any one time. And the keyboard can be assigned, note by note, to any one voice, so if you seek the ultimate, you divide it electronically into four sections, apportioning the necessary number of keys to each. Then you figure out what effects (as described above) you want to be handled by such factors as key velocity, and what are better controlled by foot switches and pedals. The basic ADS 200 has two outputs, so the performance can be panned as it plays. Perhaps more importantly, the action of the foot switches can be programmed to sequence in any of the other voices stored in disc memory. They pop up on the keyboard with a speed that is one of the most attractive features (and a trade secret) of this manufacturer. This capability is labeled "Ensemble" on the synthesizer's control panel, and eight buttons do it all, plus a good deal more that there's no space to describe.

Once the performance has taken place (and of course you don't need Horowitz to perform it; you can go back, patch up, alter, make insertions, and indulge in other almost obscene liberties), it is directly retrievable from the floppy disc that received it, whether it was simultaneously sent out to a studio recorder or a live audience or not. You can get it printed out as parts on charts, and you can transfer any part to another voice, to see if it sounds better. You can alter it in pitch and overall tuning to agree with other instruments that might be mixed with it. The same goes for tempo.

THE VALLEY OF THE DOGS

The Fairlight CMI is not based on digital oscillators, but employs a free-floating record/reproduce system that is coordinated by a variable sampling rate. Innate storage capacity, in seconds, depends crucially on the complexity of the

Figure 2. The Con Brio's CRT performs a variety of functions, including (A) the display of real-time notation for keyboard performance, and (B) a read-out of digital files available to the user.

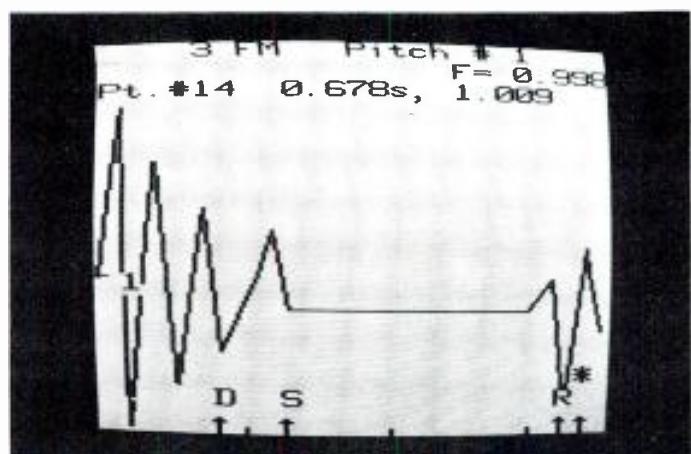
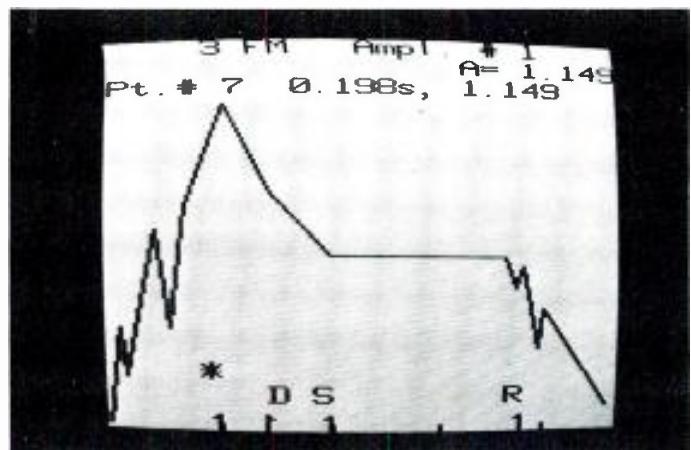
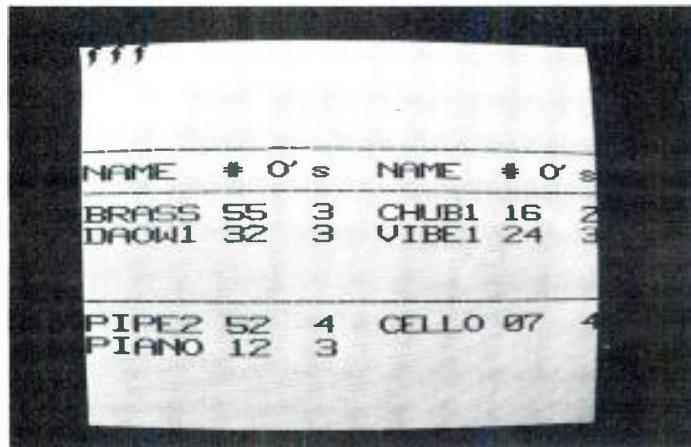
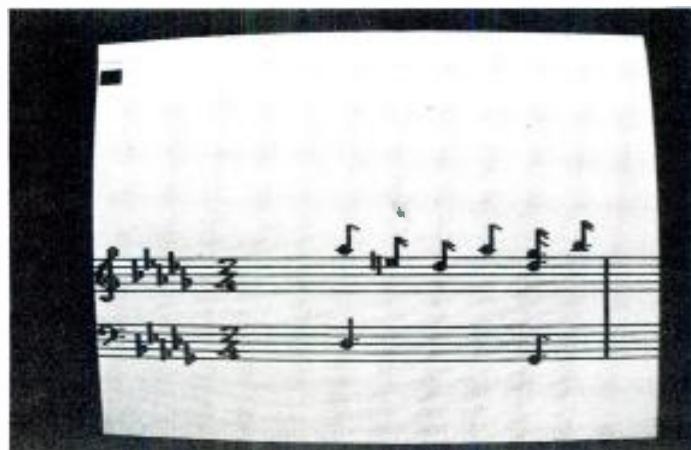


Figure 3. CRT displays of amplitude (A) and frequency (B) envelopes.

input, but most of the storage takes place on floppy disc anyway, so there is no cause for concern.

Two disc drives are incorporated in the main CMI unit. One holds the disc that regulates the control functions of the system, and is a permanent fixture (ROM, in other words). The other is the field on which you play your games.

One of the more impressive games the CMI can play is the recording of a few seconds of any sound or sound-like electronic signal, which can then be altered as desired and thrown onto a keyboard to become music. When I visited the Santa Monica distributor for this Australian-made device, there proved to be some dog barks stored in memory for experimentation. And since Santa Monica is not too many feet above sea level, and many surrounding parts of Los Angeles are... well, valley of the dogs.

This is not at all meant as a denigration. The Fairlight CMI is a fascinating creation. But like the Linn LM-1, it probably doesn't qualify as a true synthesizer, because it is silent unless it has received an input at one time or another. Once it has the input, however, the things it can accomplish are truly arresting. A waveform appears on a CRT screen (625-line base) and you put a light pen to it to perform virtually any alteration you like. This is just the beginning. As you begin to shape the dynamic and harmonic evolution of that waveform over time, the CRT display assists with readouts of up to eight harmonics graphed as amplitude-against-time, 3-D representations of spectral alterations with decay, bar graphs (alterable also by light pen) of relative overtone strengths, and a wealth of codes, instructions, and sequencing signals. You can literally make the smashing of a beer bottle sound like an operatic soprano, if you're willing to work at it and to develop the sense of eye-ear coordination that seems to come from experience with this technology. Furthermore, the CMI would seem to offer the chance of replicating, convincingly, any sound you could come across. It does not just switch and stack oscillators, but instead

Perfect Timing



Figure 4. The LM-1 Drum Computer

gives you intimate control over the actual waveform, which means that you can adjust for the vagaries of relative phase, studio/room acoustics, unknowns in the aborning science of psychoacoustics, and all the rest. Approaching the project as a programmer, it might take you a year to work your way through even one movement of the Bartok Fourth String Quartet as performed by the ideal ensemble, but in theory you could do it.

BACK TO THE RANCH

There's no need to point out that the Fairlight CMI potentially provides all the storage, retrieval, and performance capabilities expected of a computer-based synthesizer, and as handily as most of the rest. But larger questions loom: What if one of these things actually turns up at your studio door? What if the musician/engineer/programmer drops dead in the middle of a session? What if he thought he knew how to retrieve all inputs but has suddenly discovered that he doesn't?

There really is no question about what to do. Call a person as close to the manufacturing source as possible. Waste no time with local technicians, who will have to spend hours, or even days, puzzling out the control logic before they can begin to do useful work. These things are new and unknown in the field. If minutes count, allow 30 of them for experimentation and then get the advice of someone who really knows. The practical synthesizers have safeguards against the loss of vital information, but those very safeguards will block you from haphazard access. And because a modern synthesizer may very well arrive with the best part of a rhythm track, if not more, already loaded up in memory, extreme care is the best policy.

Should you otherwise be afraid of synthesizers? The best advice I've gotten indicates not. In the hands of an extremist, a synthesizer can zap you with a very heavy level of near-ultrasonics before you can catch your breath. But the digital synthesizers could not go truly ultrasonic even if they wanted to, because of low-pass filters at their outputs that limit them to 16 kHz or less. And no one is much interested in sound compositions that rise that close to extreme levels at inaudibility anyway. Modern synthesizers have too much to offer in the middle of the audio band, so there's no reason to press beyond it.

Digital synthesizers are usually set up so that they can lock onto an externally-generated time code, or impose such a time code on external equipment. They are quite flexible, and generally work well with such a universal standard as SMPTE time code. If you have any doubts about a particular interface, a few quick questions to an authoritative source should clear them up right away. Beyond that, I have no more advice about synthesizers except for a statement by Roger Linn: "Have fun with them. They're made to make people happy." ■

If You've Wanted To Put Time Code On Your Tape, But Can't Afford The High Cost Of SMPTE:

The ES 280 TIME CODE GENERATOR/READER Is For You!



The new, low-cost way to provide time information on all your tapes, ES 280 generates a time code which includes DAYS, HOURS, MINUTES, SECONDS AND TENTHS OF SECONDS.

With the flip of a switch, ES 280 becomes a reader, to locate that certain place on the tape.

An Amplitude Control knob is located on the rear, so that you can use the 280 with any recorder, Video or Audio.

The 280 uses tones that are compatible with the cue tones used in cartridge recording.

ES 280 COSTS \$500.00

ES 281 (READER ONLY) \$375.00

Contact us or one of our dealers, and we'll tell you how you can get your own 280.



(213) 322-2136
142 SIERRA STREET • EL SEGUNDO, CA 90245

Circle 42 on Reader Service Card

DIRECTORY OF ELECTRONIC MUSIC SYNTHESIZER MANUFACTURERS

Arp Instruments, Inc.
45 Hartwell Avenue
Lexington, Massachusetts 02173
(617) 861-6000

Buchla and Associates
Box 5051
Berkeley, California 94705
(415) 452-4136

Con Brio
975 San Pasqual Street
Suite 313
Pasadena, California 91106
(213) 795-2192

Crumar Synthesizers
c/o Music Technology, Inc.
105 Fifth Avenue
Garden City Park, NY 11040
(516) 747-7890

EMS Synthesizers
c/o EMSA
269 Locust Street
Northampton, Massachusetts 01060
(413) 586-3777

Fairlight Instruments USA
1610 Butler Avenue
West Los Angeles, California 90025
(213) 478-2414

Korg Synthesizers
c/o Unicord Inc.
75 Frost Street
Westbury, New York 11590
(516) 333-9100

Linn Electronics, Inc.
3249 Tareco Drive
Hollywood, California 90068
(213) 850-0741

Moog Music, Inc.
2500 Walden Avenue
Buffalo, New York 14225
(716) 681-7242

Oberheim Electronics, Inc.
1455 19th Street
Santa Monica, California 90404
(213) 829-6831

PAIA Electronics, Inc.
1020 West Wilshire Blvd.
Oklahoma City, Oklahoma 73116
(405) 843-9626

Roland Corp. US
2401 Saybrook Avenue
Los Angeles, California 90040
(213) 685-5141

Sennheiser Electronic Corp.
10 West 37th Street
New York, NY 10018
(212) 239-0190

Synare Electronic Drums
c/o Star Instruments, Inc.
P.O. Box 145
Stafford Springs, Connecticut 06076
(203) 684-4258

Synton Electronics BV
Zandpad 46, Postbus 83
3620 AB Breukelen, Holland
03462-3499

360 Systems
18730 Oxnard Street
Suite 215
Tarzana, California 91356
(213) 342-3127

Yamaha International Corp.
P.O. Box 6600
Buena Park, California 90622
(714) 522-9105

MAC 81 *Electronic Modification*

Saturday
April 25, 1981
Hermann Hall
Illinois
Institute of
Technology
Chicago, Illinois

of
Musical
Sounds

- ✖ *Characteristics of Musical Sounds*
- ✖ *History of Electronic Modification*
- ✖ *Time Domain Modification*
- ✖ *Frequency Domain Modification*
- ✖ *Non-Linear Modification*
- ✖ *Synthesizers*
- ✖ *Future Trends*

The 67th AES Convention

MORE THAN 6000 visitors jammed New York's Waldorf-Astoria Hotel last fall for the Audio Engineering Society's 67th convention (31 October-3 November, 1980). In addition to the traditional fare of technical papers and equipment exhibits, the 67th featured a nine-session workshop series which included digital editing, video for audio, and microphone techniques. And, to help answer the recurring question, "How do I get started in professional audio?", one of the workshops was an educational fair, at which "A gathering of representatives of universities and educational institutions offering courses in audio (were) on hand for personal discussions with prospective students. Information (was) provided on entrance requirements and curriculum, with details of special courses available of interest to those entering or involved in the audio field." (From the AES convention program. For a copy of the AES's Directory of Educational Institutions, write to the Audio Engineering Society, 60 East 42nd Street, New York, NY 10017.)

The convention saw the introduction of another volume in the AES's anthology series. Edited by Gotham Audio president Stephen Temmer, the 520-page volume is entitled "Disk Recording—Volume 1: Groove Geometry and the Recording Process." This brings the AES anthology series up to five volumes. The other four are: "Loudspeakers," "Microphones," "Quadraphony," and "Sound Reinforcement." The anthologies are available from the AES at \$22 a volume (\$19 for AES members).

THE EXHIBITS

179 companies jammed the hotel's grand ballroom and its balconies, several large meeting rooms, and two floors-worth of guest rooms. It has become somewhat of a tradition for at least one company to pull off a little surprise by unveiling a totally-new product at the show. This time around, it was MCI's turn—almost. Almost, because the Fort Lauderdale-based company brought along its first PCM tape recorder, but kept it in a secluded location where it was shown by invitation-only. Needless to say, we weren't invited, so all we can report is that a small quantity of machines have already been shipped to a handful of clients here and abroad. For more details, contact MCI directly.

Otari, not quite as shy, showed off its new pro'MTR-10 series of two- and four-channel recorders on the main ballroom floor. Replaceable head stacks allow the four-channel machine to be converted to two-channel operation on either quarter- or half-inch tape. The "bells-and-whistles" include a dual real-time counter, internal multi-frequency sine/square wave generator, optional tape locator with ten-position memory, and, of course, a micro-processor based servo transport system. Projected deliveries should begin at about the time you read this convention report.

At Panasonic, projected deliveries of the company's latest black box are still off in the future somewhere. Ambiguously dubbed a "Localization Processor," this second-generation prototype jolted at least a few of the golden-ears crowd, as they casually manipulated the quad pan-pots on the processor's front panel. What's so exciting about a quad pan-pot? In four-channel sound, it's certainly no big deal, but, in two-channel

systems...? It turns out that with only two output channels, the Localization Processor creates clear images in the entire frontal half-plane. Having followed the R&D work on the project for several years, this reporter found it great fun to watch people search for those side-placed phantom speakers, which of course didn't exist! (For an early report on this project, see "The Shape of Things to Come" in the June, 1979 db—Ed.)

OTHER SIGNAL PROCESSORS

At Gotham Audio, a second-generation digital reverberation system—the EMT 251—was demonstrated. If you liked R2D2, you'll love the EMT 251, which—as Gotham points out—retains the "space machine" look of the earlier model 250. But beauty (?) is only skin deep (better make that "front-panel deep"). Underneath, it's all business, with twice the memory capacity of the 250, plus self-diagnostic and internal error correction systems. The 251 provides three delay lines, a reverb cluster just prior to the onset of reverberation, and the reverberation program itself. All this is assignable across the stereo output.

The 251 is compatible with console automation systems such as the Harrison Auto-Set, and others. In other words, control-setting variations may be recorded on a data track or floppy disk, along with the usual console fader information. Hollywood's Sunset Sound Studios has taken delivery of the first 251, but you can have your own for just \$20,000.

The dbx 900 series is a modular signal processing system which now comprises five plug-in components. These are: a de-esser, a compressor, a noise gate, a parametric equalizer, and a flanger. Additional modules are on the drawing board, and up to eight will fit in a 5½-inch rack mount mainframe.

Briefly summarizing the 900 series features: the 902 de-esser can be used either as a conventional broad-band processor, or as a high-frequency attenuator; the 903 compressor features the dbx "over easy" system, and provides compression ratios from 1.1:1 to ∞ :1; the threshold range of the 904 noise gate is from -40 dB to +10 dB, with attenuation continuously variable over a 60 dB range; the 905 flanger provides delays up to 40 ms, for flanging and doubling effects. In addition, there are front-panel controls for feedback, dry/delay mix, depth, sweep speed, and phase reversal.

In 1969, Allison Research introduced the original KEPEX (KEyable Program EXpander). The KEPEX II is now manufactured by Valley People, Inc.—the recently-formed company that merged Allison with Valley Audio. K-II's expansion rate is variable from 1.1:1 to 100:1, with an attenuation range of up to 100 dB, and a threshold that is variable between -40 dBv and +20 dBv.

Gain Brain II is an update of the earlier Allison model 700 Gain Brain. In addition to the customary limiter/compressor functions, GB-II features a switchable linear/logarithmic release time, and a "ducker" function. This allows up to 48 dB of attenuation in the normal signal path. The attenuation is regulated by the presence of a signal in a side chain. Thus, a narrator's voice can be heard over a music program, or an interactive vocal-instrument balance can be set up.

Valley People also offers consultation services, including LEDE acoustical design and TEF measurements. (LEDE and



Otari's MTR-10 series of professional tape recorders is available in two- and four-channel formats. The MTR-10-4 is seen here.

TEF are registered trademarks of Synergetic Audio Concepts.) In addition, the company provides systems engineering, installation and construction services.

AT THE MIKE SHOP

"The Great British Spring" has nothing to do with the weather: it's a six-spring reverberation system available at the Mike Shop with (\$624) or without (\$499) a two-band parametric equalizer. The Mike Shop also markets the BEL BC3 Noise Reduction System. According to the spec sheet, the BC3 is dbx-compatible, with separate encode and decode circuits, thereby eliminating the need for record/playback switching. An eight-channel system costs \$1,575.

The Mic Mix Company's new XL-500 Master Room is an elegant three-piece reverberation system. The main control system is a 5½-inch rack mount unit, containing left- and right-channel equalization, chamber mode selector (plate, room, hall) and decay and mix controls. Chamber mode and decay time, as well as equalization bypass, may also be adjusted from a small remote control box located near the operator. The remote control is activated by depressing any of its mode switches. Decay time and chamber drive readouts are located on both the main control unit and the remote controller.

MICROPHONES AND ACCESSORIES

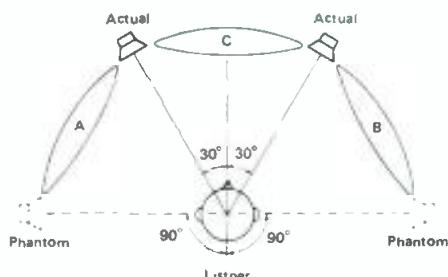
AKG's new C-414EB/P48 is being publicized as a "studio microphone for digital recording." Although its output is still analog, AKG states the 414EB is equal to the "pristine transmission qualities of digital technology." This means an equivalent noise level of 18 dB SPL (DIN 45 405), and a dynamic range of 124 dB. At 142 dB SPL, the THD is 0.5 percent. A four-position switch selects omni-directional, cardioid, hyper-cardioid, or figure-8 patterns.

Audio-Technica has introduced six new electret microphones and a battery power supply. Four of the mics are in A-T's "Artist Series": the ATM 10R (omni) and the ATM 11R, ATM 31R and ATM 91R (uni). The spec sheets identify the 10R and 11R as instrument mics, and the 31R and 91R as vocal mics. In



Panasonic's second-generation prototype Localization Processor. Once in production, the control section will be a considerably smaller package, with most of the electronics located in a remote rack mount location.

The Localization Processor gives the operator control over panning assignments within the entire frontal half plane. This includes depth, as well as angular positioning.



Audio-Technica's regular series, the AT 803R is a sub-miniature omni-directional design, and the AT 813R is a standard-sized uni-directional microphone.

Panasonic's expanding RAMSA series now includes the WM-8100, a uni-directional back-electret condenser microphone, with low-cut, high-cut and sub-sonic filter switches, and a floating suspension system. A built-in LED indicates battery voltage condition. When an external power supply is connected, it overrides the internal battery.

Audio Engineering Associates' MS-38 Matrix Decoder is used for recovering left- and right-channel information from M-S (Mid-Side) stereo signals. The decoder accepts line-level M-S inputs, and is designed for insertion in the signal path immediately after the microphone preamplifiers. Alternatively, it can be used for decoding two-channel M-S tapes.

And back to the Mike Shop again. In addition to selling Britain's Calrec microphones, the company imports Musiflex mic cable. The cable is sold in 100-foot lengths and is available in eleven colors. A conductive thermoplastic shield takes the place of the customary braided wire or metal foil, which greatly simplifies the task of preparing the cable for plugs.

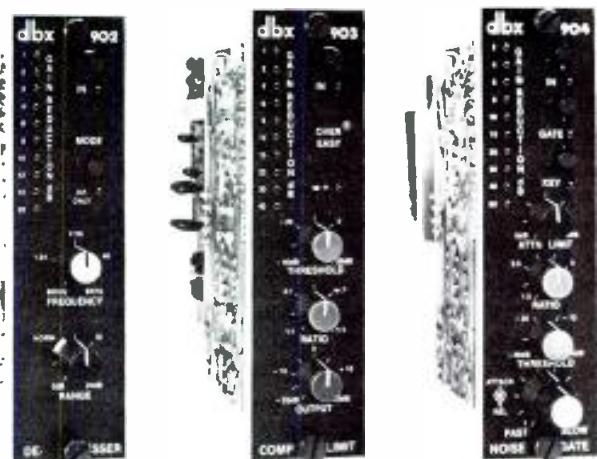
ODDS-N-ENDS

FAX Audio's Series 1 programmable fader combines a series of eight recessed membrane switches, with a Penny and Giles conductive-plastic linear-motion fader. A vertical column of 30 LEDs simultaneously indicates the levels of both read (dim LED) and write (bright LED) functions. The membrane switches control the read, write and update modes, as well as trim, mute and solo functions. A seven-segment LED displays group assignment, which is also controlled by a membrane switch.

Eventide Clockworks real-time spectrum analyzer is a plug-in card system for use with a Commodore PET, Radio Shack TRS-80 or Apple computer. For under \$600, it turns your personal computer into a spectrum analyzer, with 31 one-third octave bands. The display can be either linear or logarithmic, and decay and gain are varied by keyboard commands. ■



The EMT 251 second-generation Digital Reverberation Unit.

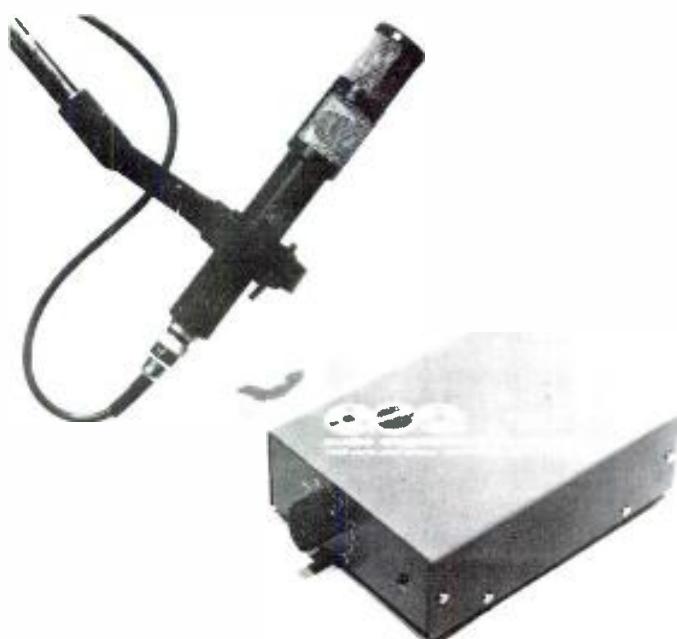


The WM-8100 is the latest addition to Panasonic's RAMSA line of pro' audio products.



Audio-Technica's sub-miniature electret condenser microphone.

This Audio Engineering Associates photo shows the MS-38 Active M-S Matrix System. Also seen is AKG's C34 stereo microphone.



db Classified

Closing date is the fifteenth of the second month preceding the date of issue.
Send copies to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road, Plainview, New York 11803

Minimum order accepted: \$25.00

Rates: \$1.00 a word

Boxed Ads: \$40.00 per column inch

db Box Number: \$8.50 for wording "Dept. XX." etc.

Plus \$1.50 to cover postage

Frequency Discounts: 6 times, 15%; 12 times, 30%

ALL CLASSIFIED ADS MUST BE PREPAID.

Lexicon Prime Time: FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

AUDIOTECHNIQUES—Red Series Time Sync. Biamplication and time synchronization for Altec 604 monitors. \$695 new, \$475 demo unit. **BILL LEVY, Audio Contractors Inc., Buffalo, NY, (716) 856-7717.**

SCULLY, NEW and used: FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

AMPEX, OTARI & SCULLY recorders in stock for immediate delivery; new and rebuilt, RCI, 8550 2nd Ave., Silver Spring, MD 20910. Write for complete product list.

COMPLETE 8-TRACK RECORDING PACKAGE using Otari MX-5050-8, Sound-workshop 1280B-8EQ. Reverb, comp/limiters/effects, microphones, stands & cables. A complete studio package with one year's use, excellently maintained and with many extras. Write or call for complete information. **Michael Ebert, Box 364, Peru, Ind. 46970, (317) 473-3290.**

2 AMPEX 440-C 1" 8-track tape machines, one complete, one missing 4 tracks of electronics, call **Milton at (809) 774-7341.**

dbx 155: FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

AMPEX AG-350-2: consoled, \$1300.00; unmounted, \$1100.00. Crown 800 transports-quad heads, \$500.00; with electronics, \$650.00. Magnecord 1028-2 \$225.00; no electronics \$75.00. (215) 338-1682.

PRO-SOUND equipment. Specializing in phone and mail orders. Free discount catalog. Write or call **Sonix Co., Dept. D, P.O. Box 58, Indian Head, MD 20640 (301) 753-6432.**

REELS AND BOXES 5" and 7" large and small hubs, heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116 (214) 296-2773.**

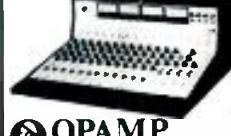
101 RECORDING SECRETS MOST ENGINEERS WON'T TELL, \$7.95 **Tune-tronics, P.O. Box 55, Edgewater, N.J. 07020.**

THE LIBRARY... Sound effects recorded in STEREO using Dolby throughout. Over 350 effects on ten discs. \$100.00. Write **The Library, P.O. Box 18145, Denver, Colo. 80218.**

COMPLETE MCI 16 TRACK STUDIO and complete color T.V. production studio for sale. All or individual pieces including choice real estate. Call for details (608) 837-4905.

AMPEX, OTARI, SCULLY—In stock, all major professional lines, top dollar trade-ins; 15 minutes George Washington Bridge. **Professional Audio Video Corporation, 384 Grand Street, Paterson, New Jersey 07505. (201) 523-3333.**

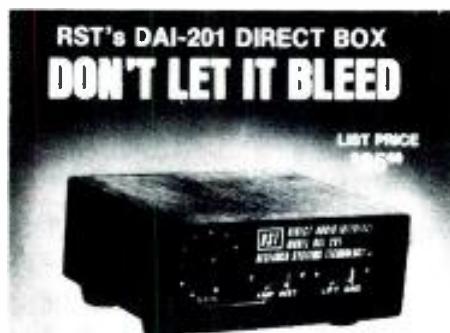
FREE CATALOG & AUDIO APPLICATIONS



CONSOLES
KITS & WIRED
AMPLIFIERS
MIC., EQ, ACN, LINE,
TAPE, DISC, POWER
OSCILLATORS
AUDIO, TAPE BIAS
POWER SUPPLIES

1033 N. SYCAMORE AVE.
LOS ANGELES, CA. 90038
(213) 934-3566

OPAMP
LABS INC.



CANADIANS!

Interested in semi-pro
multitrack recording?

Headquarters for

OTAKI

MIS

SOUNDMAN

dbx

TEAC TASCAM

DeltaLab

JBL

and many more!

Write or call



RICHARD AUDIO INC.
1078 Sherbrooke St. W. Montreal Quebec H4A 1Y1
(514) 481-4950

FOR SALE: B & K SLM Package. Includes the following; #2203 SLM, #4134 Mic., #1613 Oct., #1616 1/3 Oct., #4320 Cal., Fibre Case, Extenders and many cables. First \$4250.00 takes all. Contact: Richard Lee, Valley People, Inc. (615) 383-4737.

AMPEX MM1100/1200 24 track head assembly. Complete with Ampex heads. \$3,250/offer. Dennis (415) 494-9344.

TELEX COPIER I, COPIER II, high-speed cassette duplicators, excellent condition. \$1200.00 for pair. (212) 762-7112.

BGW: FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

JFET TUBE REPLACEMENTS for first playback stages in most Ampex Professional audio tape recorders/reproducers available from VIF International, Box 1555, Mountain View, CA 94042. (408) 739-9740.

ORBAN. All products in stock. FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

THE LEGENDARY AKG C24 Tube stereo microphone. Brand new, never used. In special custom made carrying case. \$2500.00. Call (813) 447-1487, (813) 461-2764.

UREI: FOR IMMEDIATE DELIVERY most items. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

PROFESSIONAL AUDIO EQUIPMENT

Shop for pro audio from N.Y.'s leader, no matter where you live! Use the Harvey Pro Hot-Line. (800) 223-2642 (except NY, AK, & HI) Expert advice, broadest selection such as. Otari, EXR, Ampex, Tascam and more. Write or call for price or product info:

Harvey Professional Products Division
2 W. 45th Street
New York, NY 10036
(212) 921-5920

AKG BX20 reverberation and C414 microphones. FOR IMMEDIATE DELIVERY. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

SMPTE TIME CODE BTX TELCOM: Both used and new generators, readers, synchronizers. Contact DYMA ENGINEERING, PO Box 1697, Taos, N.M. 87571. (505) 758-8686.

MIDAS FOR SALE. New 36 x 8 PRO4A CAS. (305) 961-7420, 1 PM-6 PM.

AKG, E/V, Sennheiser Shure, Neuman; FOR IMMEDIATE DELIVERY most models. UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.

SERVICES

CARTRIDGE RELOADING. Factory authorized rebuild service on Aristocart, Capitol/Audiopak A-2, AA-3, Fidelipac Cartridges. Fast turn around. Broadcast Cartridge Service, 15131 Triton Lane, #108, Huntington Beach, CA 92649. (800) 854-6419 or (714) 898-7224.

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. IEM, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.

ACOUSTIC CONSULTATION—Specializing in studios, control rooms, discos. Qualified personnel, reasonable rates. Acoustilog, Brüel & Kjaer, HP, Tektronix, Ivie, equipment calibrated on premises. Reverberation timer and RTA rentals. Acoustilog, 19 Mercer Street, New York, NY 10013 (212) 925-1365.

VIF INTERNATIONAL will remanufacture your Ampex or Scully (Ashland/Bodine) direct drive capstan motor for \$200. Average turn around time—2-3 weeks. For details write PO Box 1555, Mtn. View, CA 94042, or phone (408) 739-9740.

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAEKO, Grampian. Modifications done on Westrex. Quick turnaround. New and used cutterheads for sale. Send for free brochure: International Cutterhead Repair, 194 Kings Ct. Teaneck, N.J. 07666. (201) 837-1269.

JBL AND GAUSS SPEAKER WARRANTY CENTER

Fast emergency service. Speaker reconditioning and repair. Compression driver diaphragms for immediate shipment. NEWCOME SOUND, 4684 Indianola Avenue, Columbus, OH 43214. (614) 268-5605.

Keep
db
in
sound
order!

Special binders now available.

All you regular db readers who, smartly enough, keep all your back issues, can now get our special binders to hold a whole year's worth of db magazines in neat order. No more torn-off covers, loose pages, mixed-up sequence. Twelve copies, January to December, can be maintained in proper order and good condition, so you can easily refer to any issue you need, any time, with no trouble.

They look great, too!

Made of fine quality royal blue vinyl, with a clear plastic pocket on the spine for indexing information, they make a handsome looking addition to your professional bookshelf.

Just \$7.95 each, available in North America only. (Payable in U.S. currency drawn on U.S. banks.)

Sagamore Publishing Co., Inc.
1120 Old Country Road
Plainview, NY 11803

YES! Please send _____ db binders
@ \$7.95 each, plus applicable sales
tax. Total amount enclosed \$ _____

Name _____

Company _____

Address _____

City _____

State/Zip _____

db People/Places/Happenings

• More than 400 commercial radio stations and radio-related companies joined the **National Radio Broadcasters Association** during 1980, according to **Abe Voron**, NRBA executive vice president. This boosts total membership in the full service radio-only organization to more than 1500. The 412 new NRBA members include independent and group-owned stations from 47 states, the District of Columbia, Puerto Rico and Canada. Of the 342 new commercial stations, 149 are independent and 193 group-owned, while the 70 new associate members are radio-related manufacturers, distributors, suppliers, consultants and educational radio stations. States attracting 10 or more new members were: Alabama, California, Florida, Illinois, Indiana, Massachusetts, Michigan, Mississippi, New York, Ohio, Pennsylvania, Texas, Virginia, and Wisconsin.

• **John R. Saul**, President and founder of **MICMIX Audio Products, Inc.** died on January 2, 1981 at the age of 49. Mr. Saul was a member of the **Audio Engineering Society**, **National Association of Broadcasters**, **Society of Broadcast Engineers**, and the **Society of Motion Picture and Television Engineers**. Mr. Saul worked as a senior project engineer for **LTV Corporation** and resigned in 1972 after 20 years of service. In 1972, **MICMIX Audio Products** was incorporated, and Mr. Saul assumed the position of president.

• Late this past December, the **Los Angeles Audio Engineering Society** Chapter held an informal meeting for its members with **Mr. Bill Putnam**. Mr. Putnam, chairman of the board of **United Recording Corporation**, reflected on the early days of the record business. His talk included how he recorded some of the music giants of the past: **Ellington**, **Sinatra**, **Patty Page**, etc. Putnam fielded many questions from the audience and showed the first module from an early recording console he designed. Recognized as one of the recording industry's founding fathers, Bill Putnam was responsible for the creation of United Recording, now **United/Western** located in Hollywood; the formation of **U.R.E.I.**—an industry manufacturing leader in studio equipment; **Coast Recorders** of San Francisco, and an information systems firm named **Teletronix**.

• **Empirical Audio** of Ossining, NY, is pleased to announce the sale and installation of a new **Trident TSM Automated Recording Console** to **MasterSound Productions**, of Franklin Square, NY. According to **Winn Schwartau**, president of Empirical Audio, MasterSound's Trident console is the largest board delivered to date. The custom TSM has 48 inputs, with 32 monitors, giving the board a full 80 channel remix capability. MasterSound is automating the console with a **Valley People 65K** programmer, and feeding the Trident console with **Ampex MM1200** 24 and 16 track tape machines.

• The **Recording Industry Management** program at **Middle Tennessee State University** has purchased a **Harrison Systems 2824** series console. The console, along with an **Allison 65K** programmer, was installed in the **RIM** studio by **Studio Supply** of Nashville. Recording Industry Management is a four year degree program covering all aspects of the recording industry.

• **Community Light & Sound** has announced that **Chris Foreman** has joined them in the newly created position director of marketing. Mr. Foreman, the former director of applications engineering at **Altec-Lansing Corp.**, is a member of **AES**, **IEEE**, **SMPTE** and is a regular contributor to many audio magazines and trade journals.

• **Charles P. Repka** has joined **RCA VideoDisc Operations Division** in Indianapolis, Indiana as a member of the Systems Engineering group. His experience includes work as a Systems engineer on the **Sky Lab** program at the **Marshall Space Flight Center** and as a **EW Systems** engineer for **I.T.T. Avionics**. In recent years, Mr. Repka has been a recording engineer for **Vanguard Records**, disc mastering engineer for **Masterdisk**, freelance recording engineer and writer on audio topics for a variety of audio journals. Mr. Repka has been an active member of the **AES**, serving as committeeman, vice-chairman and chairman of the N.Y. Section. He has also chaired the Disc Technology session at the N.Y.C. AES convention.

Mr Repka's duties at **RCA** will include the development of audio system parameters for the current and future **VideoDisc Systems**.

• **Empirical Audio** has completed negotiations with **Ran-Steele Audio** of New York City for the sale of 24 **Advanced Music Systems** Disc Mastering Digital Delay Lines. **Ran-Steele Audio** is a manufacturer and supplier of custom mastering facilities world-wide. According to **Tom Steele**, president of **Ran-Steele**, the use of Disc Mastering DDL's will soon replace the conventional cutting method of using advance heads to feed the cutter head preview signal information. The **AMS DDL** is a programmable stereo unit, with up to 1.3 seconds of delay, and a bandwidth of 27 kHz.

• **Compact Video**, Burbank, Calif.; **Rodel Audio**, Washington, D.C.; and **ITV Ltd.**, Edmonton, Canada, all have acquired **Aphex Aural Exciters** to sweeten sound for audio and video productions, announced **Marvin Caesar**, president of **Aphex Systems Ltd.** **ITV Ltd.** is utilizing its **Aural Exciter** for sound mixdown for **MCA/Philips DiscoVision** videodisks. In addition, **ABC-TV** is using an **Aphex Aural Exciter** in producing the "Lawrence Welk Show" for syndication. **Aphex Aural Exciters** are also being used in recording studios and by concert artists and rock singers on tour. Artists using the sound enhancer in concert or on albums have included **Willie Nelson**, **Linda Ronstadt**, **Diana Ross**, **the Cars**, **the Bee Gees**, **Fleetwood Mac**, **Elton John**, **James Taylor**, **Wings**, **Donna Summer**, **Neil Diamond**, and **Barbra Streisand**.

• **Ian Robertson**, formerly engineer for **KING AM/FM**, one of Seattle's leading radio stations, is joining **Audio & Design Recording**, the UK-based Signal Processing equipment manufacturer.

• **Neve Electronic Holdings Limited** of Cambridge, England has announced the appointment of **Clary A. MacDonald** as President of **Rupert Neve of Canada Ltd.** MacDonald has been employed by Neve since March of 1974, first as Canadian technical services manager, and later as marketing manager and vice president and general manager for Neve's Canadian operations. Prior to joining Neve in 1974, MacDonald was in the service of the **Canadian Broadcasting Corporation** as broadcast engineer.

TM 500 now gives you the quickest distortion measurements ever.

Introducing the AA 501 Distortion Analyzer and SG 505 Oscillator. Fast, automated and accurate.

This new pair of TM 500 Plug-ins makes distortion measurement truly automatic to save you both time and money. For production testing, the AA 501's automatic speed provides substantial labor reduction with no loss in accuracy. Together, the AA 501 and SG 505 have the lowest harmonic distortion plus noise (THD+N) rating in the entire industry: 0.0025%.

The SG 505 Oscillator outputs a sine wave with the lowest residual distortion on today's market (.0008%). The AA 501 Distortion Analyzer uses digital processing to lock in on test signals, set levels and adjust the notch filter for nulling. All measurements, including dB levels are precalculated and then displayed on an LED readout.



For immediate action, dial our toll free automatic answering service 1-800-547-1512

The AA 501 and SG 505 are both TM 500 Plug-ins that can be installed in any of five mainframes, including rackmount, bench and portable. They can also be separated and still used as a team, even though miles apart. Or configured with over 40 other TM 500 Plug-ins currently available.

To find out more about the AA 501 Distortion Analyzer and SG 505 Oscillator, contact your local Tektronix Field Office or write Tektronix, Inc.

TM 500 Designed for Configurability

Tektronix®

COMMITTED TO EXCELLENCE

U.S.A.
Tektronix, Inc.
PO Box 1700
Beaverton, OR 97075
800 547-1512
Oregon 800 644-9051

Asia, Australia, Canada,
Central &
South America, Japan
Tektronix, Inc.
America's Pacific
PO Box 1700
Beaverton, OR 97075

Africa, Europe,
Middle East
Tektronix Int'l, Inc.
European Marketing
Center
Postbox 827
1180 AV Amstel
The Netherlands



EXCELLENCE!

The choice of the prominent studios and top artists. Neve music recording and mixdown consoles. We cordially invite you to join the Neve world of excellence. Consider the incomparable Neve Model 8108 with microprocessor controlled Central Assignment System, instant interrogation, and choice of manual, VCA or NECAm fader systems, just to name a few of the 8108's outstanding features. From 32 inputs/24 track to 56 inputs/48 track, the 8108 is the most advanced audio console ever!

Please call or write. Neve's in your future.



Neve

Rupert Neve Incorporated, Berkshire Industrial Park, Bethel, Connecticut 06801 Tel: (203) 744-6733 Telex: 969638

Rupert Neve International, 7533 Sunset Blvd., Hollywood, California 90046 Tel: (213) 874-8124 Telex: 194942

Rupert Neve Incorporated P.O. Box 120907, Nashville, Tennessee 37212 Tel: (615) 385-2600

Rupert Neve of Canada, Ltd., 2721 Rena Road, Malton, Ontario L4T 3K1, Canada Tel: (416) 677-6611 Telex: 983502
Neve Electronics International, Ltd., Cambridge House, Melbourn, Royston, Hertfordshire, SG8 5AU England Tel: (01763) 60776

Rupert Neve GmbH, 6100 Darmstadt Bismarckstrasse 114, West Germany, Tel: (06151) 81764

Circle 12 on Reader Service Card